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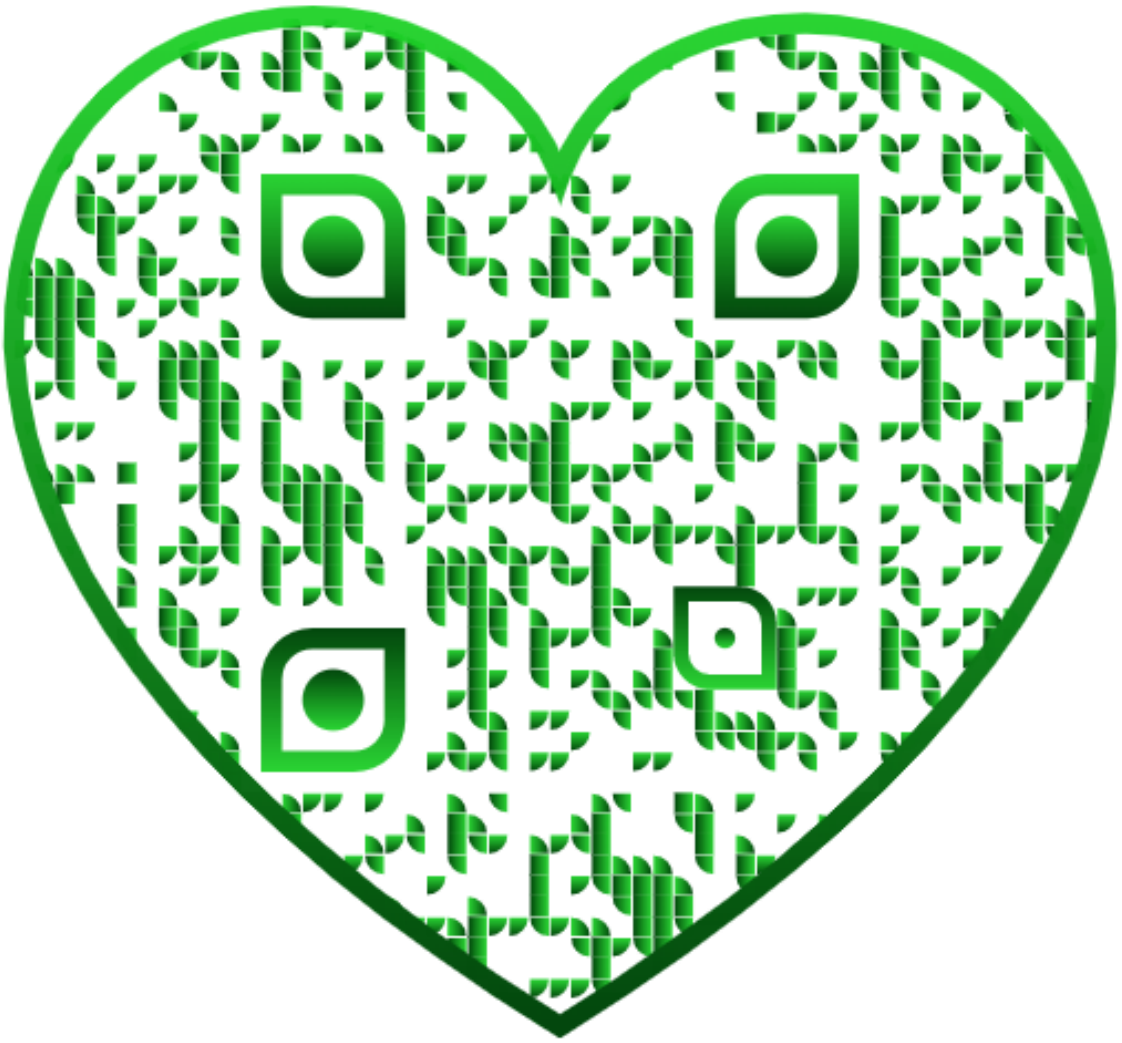

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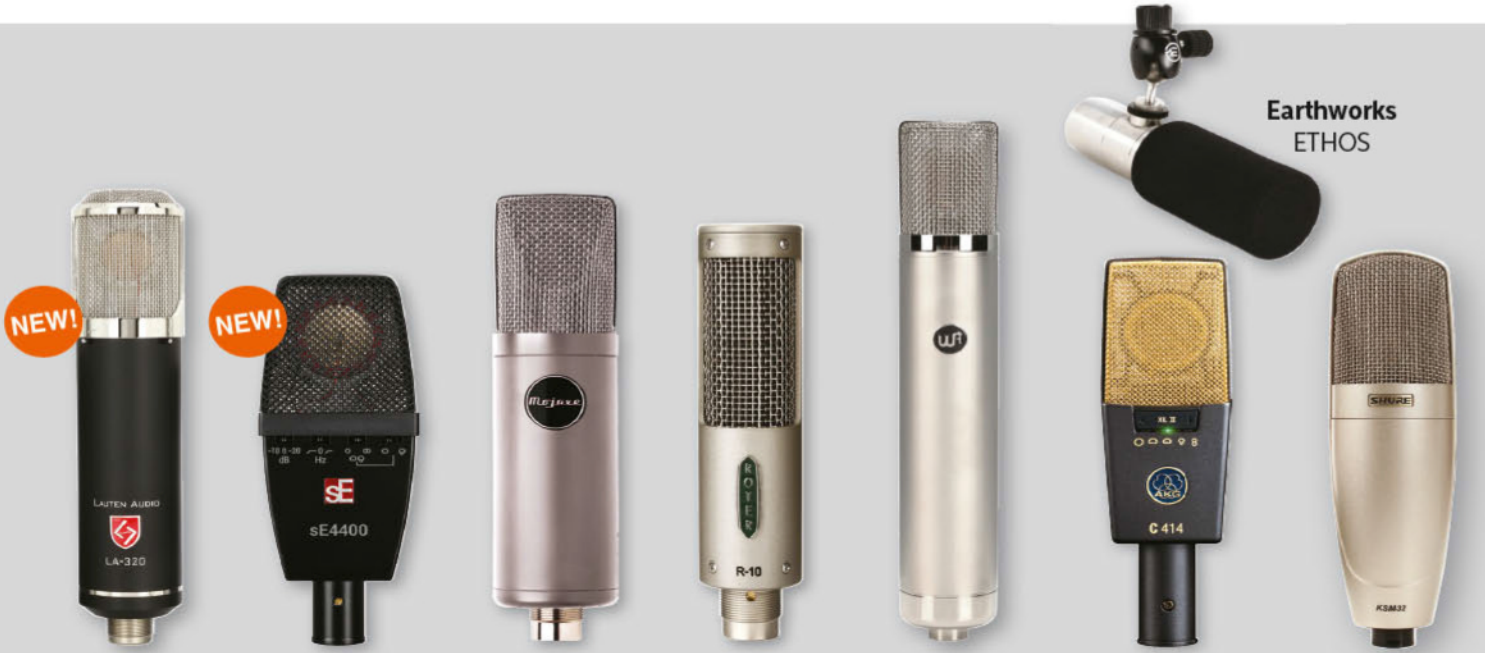

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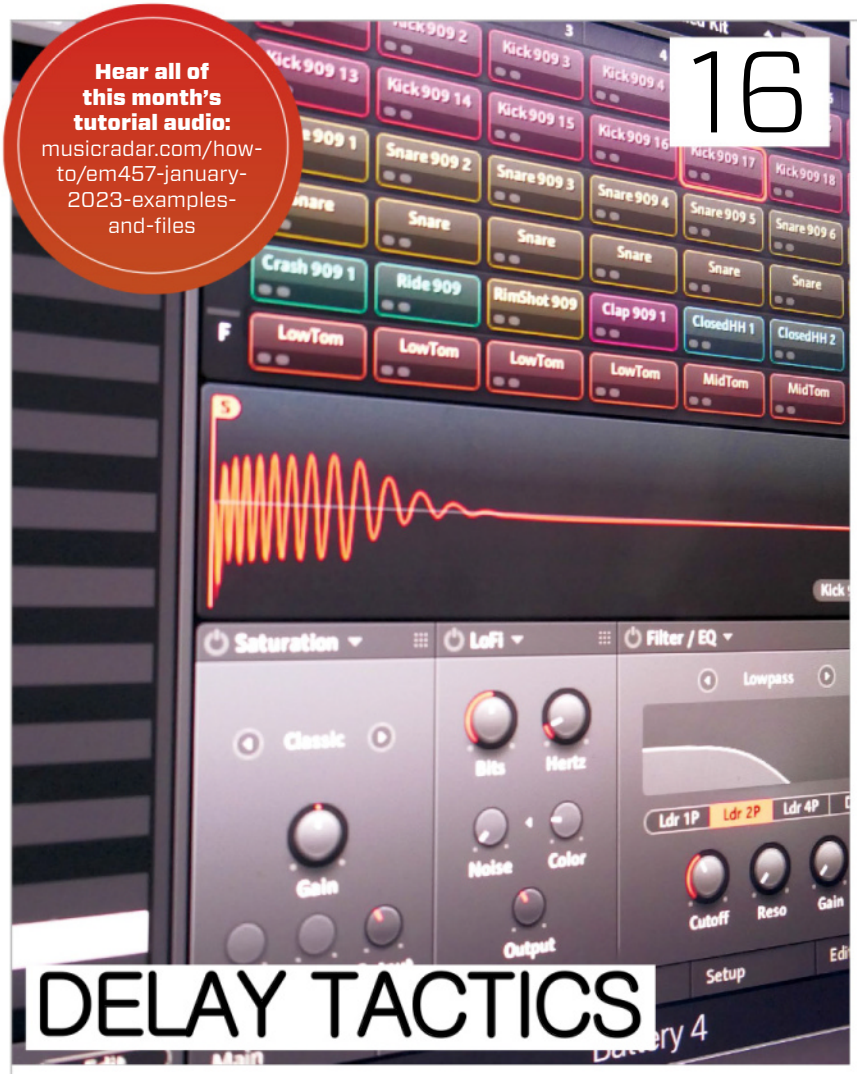
Hear all of this month's tutorial audio: musicradar.com/how-to/em457-january-2023-examples-and-files



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TURN ON INSPIRATION

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Editor's Note

A digital decade

In terms of electronic music gear, the defining trend of the previous decade was definitely a move toward affordable analog gear. In the '00s, analogue synths and effects had been largely sidelined in favor of flexible digital instruments like Korg's microKORG and MS2000, and by the end of that decade the likes of NI's Maschine were pushing us a primarily in-the-box approach to production. However, in the '10s things shifted the other way significantly, with instruments like Korg's Volca range, Arturia's Mini and MicroBrutes and Novation's Bass Station II all putting genuine analog synths into the hands of even the more cash-strapped music makers out there.

It now feels like trends have shifted once again though. It's not that there aren't plenty of great analog synths being released – from PWM's Malevolent to Oberheim's OB-X8 – but many of the most exciting new instruments we've seen in recent years have been proudly digital.

Once again, Korg are impressively at the head of the trend, with their brilliantly versatile Wavestate, Opsix and Modwave, but there's a wonderfully variety of digital synths out there right now, from Modal's Cobalt and Argon ranges to the innovative takes on wavetable synthesis used by ASM's various Hydrasynths. Arturia's new MiniFreak is the latest addition to this digital polysynth party. Read all about it in this issue's news section.

Of course, digital tech isn't confined to the hardware realm. In this issue's Masterclass feature, we're exploring some of the most cutting-edge approaches being deployed in the plugin realm.

We hope you enjoy the issue.

SI TRUSS
EDITOR-IN-CHIEF

Editor's choice

FROM THE ISSUE:
deadmau5
An undesputed icon of modern electronic music, you can always count on the mouse-headed one for a few bold opinions and ideas. In this issue's interview feature, we discover what he makes of the latest tech trends.
Review: Page 26

FROM THE ARCHIVE:
Arturia MicroFreak
Arturia's MiniFreak has caught our attention this month, but it builds on the ideas of the company's previous digi synth, MicroFreak. Catch up on what made the pintsized paraphonic synth special in our review.
See more: bit.ly/EMMicroF

AROUND THE WEB:
macOS Ventura
Apple have released a new OS, which is always tricky to navigate for music makers. At the time of writing, not all plugins are likely to be Ventura compatible. MusicRadar have more info **See more: bit.ly/EMventura**

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NEW GEAR

ARTURIA UNVEIL HYBRID MINIFREAK POLYSYNTH, PLUS A NEW INTERFACE, PLUGIN AND MORE

Arturia's October 'rendez-vous' event saw the French outfit unveiling four new releases, touching on both hardware and software. Of these, the definite highlight is the MiniFreak, a forthcoming polysynth that takes the digital-meets-analog ideas of the MicroFreak and runs with them.

MiniFreak

Despite its name, the MiniFreak isn't exactly what we'd call 'mini' – in fact, it expands on the form of the MicroFreak considerably and finds room for a 37-note slim, synth action keyboard. Just like the MicroFreak, the Mini pairs a digital oscillator section with a state-variable analog filter and some neat modulation tools.

The upgrades don't just come down to the size of the hardware though. Whereas the MicroFreak could function as a paraphonic synth to play up-to four notes, the MiniFreak boasts a full six voices of polyphony, making it a solid source of morphing digital pads and chords. In fact, the Mini has several voice modes, allowing it to be used as a monosynth, 6-voice polysynth, 12-voice paraphonic synth, or in unison mode for thick, detuned tones.

The other significant addition comes in the form of a three-slot digital effect engine. Each of these slots can be assigned one of several effect types, which include reverbs, delays, modulation effects, EQ, distortions and compression. MiniFreak's outputs are stereo, meaning the digital effects can be used to add width and stereo movement to sounds.

Like the MicroFreak, the oscillator section uses a selection of 20 digital synthesis modes, including several developed by modular brand Noise Engineering, each of which can be adjusted using Wave, Timbre and Shape parameters. The MiniFreak, however, packs two oscillator 'engines', which can be blended for multi-synthesis approaches, or used in serial, so that engine 2 can apply distortion or waveshaping approaches to the output of engine 1.

The 37-note, velocity-sensitive aftertouch keyboard looks like a significant step up from the MicroFreak's capacitive control surface, and should make it much more playable. There's also expanded sequencer/arpeggiator functionality and randomization features.

What's more, the MiniFreak even comes with a plugin version of itself. The VST version, MiniFreak V, of MiniFreak uses identical sound engines, modelled filters, presets and controls, so presets and your own sounds can be shared between this and the hardware.

MiniFreak is available to order now priced at $599.

MiniFuse 4

Alongside the MiniFreak, Arturia have added a new flagship interface to the MiniFuse range. The MiniFuse 4 is 4-in/4-out interface (the clue's in the name), that promises "best-in-class" audio quality, which equates to 110dB dynamic range and equivalent input noise of -129dB. MIDI I/O and a USB hub are also onboard, and there are dual headphone outputs.

As is becoming common these days, MiniFuse 4 has a virtual stereo loopback channel that enables you to record audio alongside the inputs for easy 'content' creation, while the bundled software includes Ableton Live Lite, Analog Lab Intro and Native Instruments' Guitar Rig 6 LE. It's available now in black or white, priced at $219.

MiniLab 3

The company have also refreshed their MiniLab controller range, now billed as "the world's first eco-designed MIDI keyboard". Arturia claim that it's made with a minimum of 50% recycled plastic and generates less waste during production, while the packaging is said to be 100% recyclable.

In terms of specs, it's a similar offering to its predecessor, giving you a compact 25-note keyboard, various other control features and sounds courtesy of the tightly integrated Analog Lab intro software that comes included.

The slim keys – which are said to feel great, by the way – are joined by pads, faders and knobs, plus touchstrips for pitchbend and modulation and a new OLED display. The faders are new to the MiniLab, though space constraints mean that there are fewer faders than on the MiniLab mkII.

There's also a built-in arpeggiator, a Chord mode and custom DAW presets for Ableton Live, FL Studio, Logic Pro, Bitwig Studio and Reason.

Augmented Grand Piano

Building on the template laid down by their Strings and Voices plugins, Arturia have released a third instrument in the 'Augmented' range that focuses on the grand piano. Like its stablemates, Augmented Grand Piano blends samples and synth engines to create cinematic sounds that are particularly suited to film scoring, video game soundtracks and library music production.

The samples are taken from an '80s Steinway grand, but aren't the simple recordings you might be used to. Unusual techniques involving bowing, ping-pong balls and tape processing were used in their creation, and that's before you even get to the multi-mode synthesis layer.

It's available now as a standalone plugin, or as part of a bundle with the other Augmented instruments.

MOOG'S MOOGERFOOGER PEDALS RETURN IN PLUGIN FORM

The demise of Moog's Moogerfooger range of effect pedals in 2018 was a moment of sadness for many people, who'd come to love this collection of high quality analog stompboxes. Now it's back, though this time, in the form of a bundle of seven plugins.

Bob Moog introduced the Moogerfooger brand in 1998, kicking things off with the MF-01 low-pass filter and MF-102 ring modulator, both of which were inspired by his synth module designs. The MF-103 phaser was released the following year, and four further products followed.

All seven Moogerfoogers have now been emulated for the DAW generation, with each plugin being based on the original analog circuits. The effects now operate in stereo and have expanded feature sets; all parameters can be automated and you can create, save and load presets.

The hardware Moogerfoogers all featured CV inputs and outputs, meaning that they could be connected together to produce inter-related effects. This feature is recreated in the software versions; each instance of a Moogerfooger can modulate the parameters of any other.

There are digital attenuverters on every CV input, along with sidechain and DC offset capabilities.

The Moogerfoogers run on Mac and Windows in VST/AU/AAX formats, and the bundle is available now priced at $249.

LOGIC 10.7.5 ADDS FREE TEMPO RECORDING, ABLETON LINK SUPPORT AND PERSONALIZED SPATIAL AUDIO

Apple have casually dropped version 10.7.5 of Logic Pro. This isn't a 'full point' release – these tend to be bigger – but there are some potentially useful new features nonetheless. Chief among these is probably free tempo recording, one of several Smart Tempo enhancements. This enables you to record audio in a new or existing project with no metronome or defined tempo, regardless of the overall project tempo mode. So you no longer need to be a slave to the rhythm.

You can add a free tempo recording button to the control bar, and when you click it, recording starts immediately, with no count-in or click. The selected track is soloed and no other sound is audible as you record. Once you've finished recording, you can decide whether you want to apply the tempo of the recorded region to the project, apply the project tempo to the region, or proceed without analyzing the region tempo or changing the project tempo.

The idea is that you can use free tempo recording to quickly capture ideas without worrying about a click, record material that doesn't have a fixed tempo or timing, or go on to make new content follow the tempo of your recorded performance in an organic way.

Elsewhere in Logic Pro 10.7.5 there's now Ableton Link support, which will make it easy to create synced jams involving other DAWs or the many iOS apps that are also compatible with the standard.

There's also personalized Spatial Audio support: if you're running macOS 13 or later, this promises "an even more precise immersive audio experience over headphones by using a customized head-and-ear-measurement profile that users can create on supported iPhones."

Other enhancements include the ability to record the output of MIDI plugins, edit the gain of audio regions with the Gain tool and use nested track stacks within both folder and summing stacks.

There are a couple of new sound packs, too – Beat Tape (Hip Hop) and Modular Rhythms (Synth Drums) – and the 35 individual stompboxes that are included in Pedalboard are now available to use as individual plugins on both audio and software instrument tracks.

The Logic Pro 10.7.5 update is free for existing users, though Apple does recommend that you backup your currently installed version and your projects as a precaution.

RNBO JUST MADE IT TEN TIMES EASIER

FOR MAX USERS TO BUILD PLUGINS, GUITAR PEDALS AND DIY HARDWARE

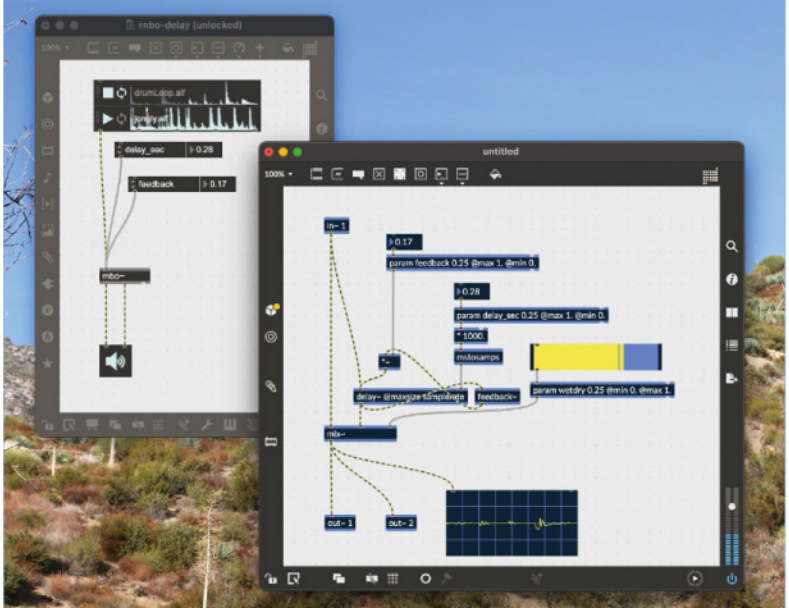
Cycling '74 have announced the release of RNBO. Pronounced 'rainbow', it's a new patching environment and visual editor that enables users to build audio patches that can be exported to Raspberry Pi, Web Export, VST/AU plugins and more.

RNBO is an add-on to Max, Cycling '74's long-established visual programming environment. The developers claim that the idea behind RNBO is to give users the best of Max – "rapid iteration and creative exploration" – along with the freedom to take their work and use it outside of Max, quickly and easily, without having to write any code.

This means patches created in RNBO can be turned into VST3/AU plugins, exported to DIY hardware devices, and built into websites with relative ease. (Cycling '74 point to Ableton's Learning Synths tutorial website as an example of a browser-based project developed in RNBO.) Patches can also be compiled into self-contained Max objects or exported as raw C++ source code. Essentially, RNBO is a potential game-changer for DIY music-makers, making it far easier to share tools, ideas and sounds they've created in Max with others, and utilize them in all-new musical contexts.

Though it's been described as an add-on to Max, RNBO has its own distinct visual editor and is based on a new engine that's been built from the ground up to enable its increased flexibility. However, RNBO's patching environment should be familiar to Max users, and makes use of many objects from the Max library. The sound users will hear while patching in RNBO is generated and compiled as code that updates in real time. This means the results will sound identical, no matter how they're exported.

RNBO is available to Max users now as a subscription, for either $10/mo or $100 a year. Alternatively, you're able to purchase it outright for $299.



AKAI RELEASE THE MPK MINI PLUS

Akai have upsized their MPK Mini MIDI keyboard and released a Plus version. This comes with an extended three-octave keyboard, a built-in sequencer, MIDI and CV/Gate connectivity, proper pitch and mod wheels, MPC pads, knobs, transport controls and more. That's a pretty comprehensive spec list, and one that would seem to put the MPK Mini Plus in direct competition with Arturia's popular KeyStep 37 controller.

The 37 slim keys use Akai's second-generation dynamic bed, while the polyphonic two-track sequencer – designed for creating melodic and drum patterns – comes with Step and Live modes. There are eight notes of polyphony per key step, 16 notes per drum step and an onboard arpeggiator.

As well as being useful for drum playing and programming, the pads can also be used for making program changes and triggering additional CC parameters. As you'd expect, the eight 360-degree rotary knobs are all assignable, to boot.

As with so many modern MIDI keyboards, the MPK Mini Plus offers Chord and Scale modes – used for triggering chords with a single key and locking the keyboard to a selected scale respectively. Both modes feature plenty of options and there are 15 scale settings.

The transport controls cover the obvious bases, and the joystick enables you to assign four parameters to the X/Y axis and morph between them. There are three CV/Gate outputs (pitch, gate and mod) and CV Clock I/O, along with USB and full-size MIDI I/O ports. You can also plug in a sustain pedal.

The MPK Mini Plus is available now priced at $169. Find out more on the Akai Pro.

SYNCLAVIER RETURN WITH THE REGEN DESKTOP SYNTH

You probably never owned a New England Digital Synclavier – an early digital synth that became a fixture in high-end studios in the '80s – but you've almost certainly heard one. It's all over *Thriller* and *Bad*, Michael Jackson's two all-conquering albums of the era, and others users included Tony Banks of Genesis, Geoff Downes of Asia, and Mute Records' Daniel Miller, whose Synclavier would be used on a few Depeche Mode productions. One particularly notable use of the Synclavier (specifically, the Synclavier II) was the intro to MJ's 'Beat It', which was recorded using one of the instrument's stock presets.

The synth is also remembered for its humongous price tag. A basic system cost $13,000, but adding more features could raise this to as much as a mind-bursting $400,000, ensuring that the Synclavier was very much a rich person's plaything.

Fortunately, this isn't quite true of the Regen, a new desktop synth bearing the Synclavier name that's just been announced. Promising to draw on "the rich history of the Synclavier, while simultaneously breaking new ground," this offers both additive and subtractive synthesis and sampling.

The Regen contains the Synclavier II DSP engine in an enhanced and optimized form. There's additive synthesis with full control of 24 harmonics, while the inclusion of subtractive synthesis means that super saws, PCM square waves, and noise generation are within reach. You can also use samples for your carrier waveform. You can mix and match waveforms, building your sounds with up to 12 partials, and crossfades can be assigned to the likes of velocity, pressure, the mod wheel and your keyboard.

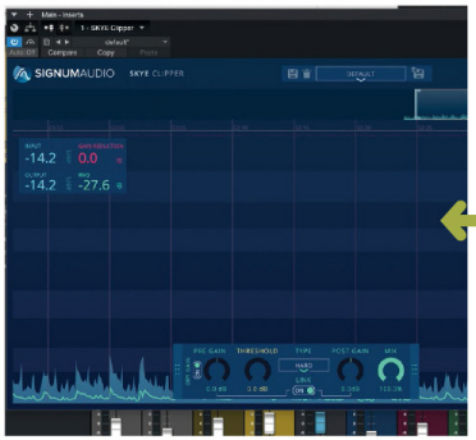
Regen comes with more than 250 so-called 'timbre presets' including six artist timbre libraries. There are also two exclusive sample libraries containing more than 300 new samples, plus a "mega library" that contains more than 700 original samples from the New England Digital era.

Control options for the Regen include a 'touch swiper' and 12 selector pads, and you can assign multiple effects to each of the 12 tracks (the synth is multitimbral or you can stack/layer sounds. Four USB ports enable you to plug in keyboards and other peripherals, and there's a further port for connection to your DAW. You can keep track of what's going on via the two high-resolution screens, and there's a "high fidelity" DC-coupled audio out (stereo XLR or TS) with 130dB of dynamic range.

Signum Audio Skye Clipper

Clipping is, technically, a time-honored staple of music production. For most of its life, though, it was unwanted – the distortion caused by a signal whose level was too high for a given piece of gear. These days, clippers and soft clippers serve a lot like brickwall limiters, stopping a signal from going over a certain threshold. The effect is useful in mastering to achieve loudness, but also tends to introduce saturation, especially when pushed hard.

Signum Audio's Skye Clipper takes on clipping with seven modes of operation, which we'll demonstrate through this article. It also boasts proprietary technology to reduce aliasing artefacts, aiming for a smoother clipping effect that can help push your signal harder and hotter. You can get Skye Clipper or a free trial version at signumaudio.com



1. Starting with Gain

Here's Skye Clipper straight out of the box. We can see our signal hiding down at the bottom, barely tickling the floor with some serious headroom above it. Sounds like a challenge. We can start by pushing up the Pre Gain parameter, which works only when the switch next to it is activated. By default, this is linked to the Post Gain knob, compensating for any change we've made here after the clipping has happened. For now, we'll switch that off to run our signal a little more healthily. At 13 or so dB, we start to hit the clipper.

3. Keeping an eye out

The graph in the top right represents your clipping transfer function, and helps you see when your signal actually gets pushed down. The metering down the right side of the plugin shows the output left and right channels (left and middle bard) as well as the amount of gain reduction coming down from the top in red (right bar). For mastering-style clipping, we may want to unlink the Pre and Post Gains to balance it ourselves; for creative, distorting clipping, we'd generally want to match the input and output meter levels.

2. Threshold operation

To do the same without changing the gain, we can simply bring the clipper's Threshold down to the input level of our signal to get things started. The clipper scalps off any transients that try to get in its way, dulling their snap a bit and keeping the dynamic range of the signal low. Bring it down lower, and the sound really starts to fuzz out. It sounds bad for most production contexts, but can really get something like a Rhodes piano going in terms of aggressive distortion. This hard clipping is definitely best for taking off the top transient peaks.

4. Clipping types

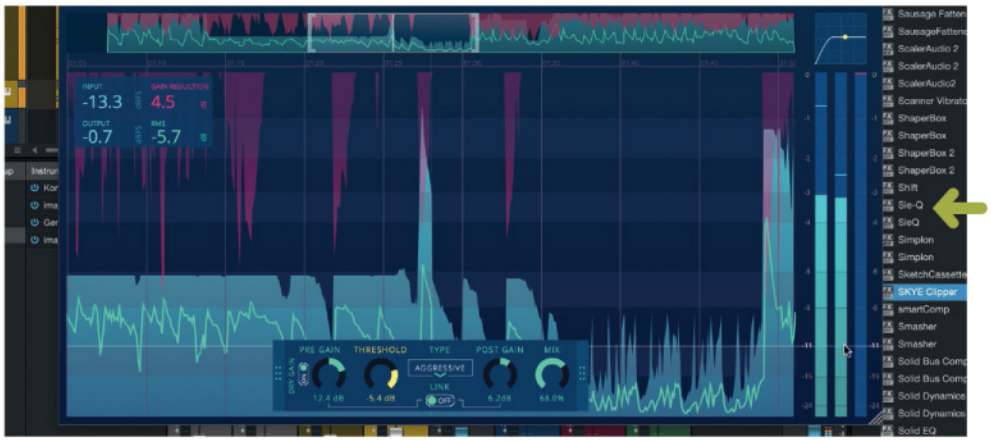
Now the moment we've all been waiting for. You can choose between the seven types of clipping (IE, seven transfer functions), using the menu in the centre of the bottom panel. Warm clipping has a nice, gentle knee shape (transfer function) that progressively clamps down harder as the signal goes higher over the threshold level. Bite straddles gentle and hard clipping, with a very similar transfer curve to the Bright setting we're about to see.

Transfer functions

Dynamics devices often contain a transfer function view. This is a representation of output against input signal, with a line plotted to determine how they connect. The most obvious example is in a compressor: with no compression applied, the line is a straight diagonal; when compression is applied, the line remains like this from its bottom left, but starts to bend at the threshold, meaning that everything over the Threshold is curtailed in level to an extent. The curve around the threshold point would imply a soft or hard knee, which represents how quickly the most extreme level of compression is reached after the threshold is breached. Clipper transfer functions work in just the same way.

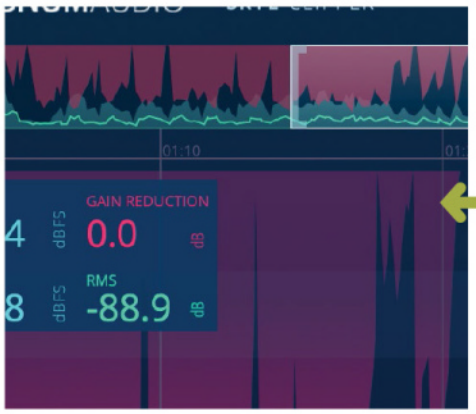
5. More clipping types

The Bright shape actually has a steeper transfer function before clipping occurs, and the clipping that does occur goes from gentle to somewhat hard quite swiftly. Bite Saturator, meanwhile, introduces a Softness control that helps you bend the curve's knee between hard and soft. There's a Smooth type that backs things off a bit, and an Aggressive one that takes things into more firefight-ish territory – although again, the result you get will very much depend on the material you put in in the first place.



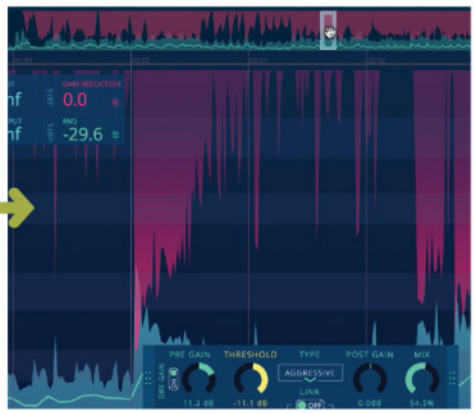
6. Mix

If you've cooked the clip very hard (a la the Aggressive distortion type), you can back things off again, while retaining some of that flavor, using the Mix control. Reducing it from 100% lets some of the dry, unprocessed signal back in. In a creative or saturation use case, the Mix control is a normal affair; in a transient-taming clipping use case, the Mix won't function much like a parallel compressor would, exactly.



7. History

Along the top of the interface, you can see back in time, checking your waveform at different points in the past, including gain reduction amounts. You can use the timeline view selector to make the viewable window smaller, taking a closer look, or larger, getting a wider overview of a large portion of your project. You can set the window back to wherever you choose, although the information will be overwritten if Skye Clipper has been active for a while longer after this was printed.



DELAY TACTICS

Delay is the pro producer's favorite effect. Here's how to get creative and cost effective with your own set of delay tactics

Read any interview with professional engineers and producers and there will no doubt be mention of their favored equipment, whether it be synthesizers, outboard equipment or more often than not these days, favored plugin effects. Dig a little deeper and you might get an insight into some of the tricks of the trade, and among those who wax lyrical about the wonders of reverb, we would wager that there are just as many who will effervesce about their favorite delay.

DAWs are packed with them, or you can get them from classic gear like tape recorders. Step outside and shout in the right location and they are there! Yes, delays are everywhere! So how do you make the most of the delay in a musical context?

That's where this feature comes in, with advice and tips to make the most of your delays. So if you've ever found yourself reaching for reverb before you add delay, we're going to use some tactics that might make you think again!

DELAY TACTICS

The delay differences

Getting tactical

One of the most common questions that arises from this topic is: what is the difference between a reverb and a delay? The two effects are heavily related, and often distinguished by their behavior, in terms of what we hear.

Let's deal with delay first of all; in our world of music production, delay is sometimes described as echo. That's a great way of looking at it. If you've ever been to a location, such as a vast chasm or room, where you can shout loudly and literally hear your voice bouncing back at you, we would generally describe that as echo. You could achieve similar results by shouting at a bare wall or window, where any other surrounding surfaces are either acoustically treated, or maybe within the confines of your home, where soft furnishing will dampen additional sound wave reflections (see **Fig. 1**). When we reach for a delay plugin, that is effectively what we're trying to recreate; the audible appearance of a sound, such as a voice, which will then come back at you as some form of repeat, and moreover, a repeat which will be entirely recognizable as something close to the original source.

The good news is that unlike standing on a precipice at the Grand Canyon, we have far more control over our echo, or delay, meaning that we can dictate and change the number of repeats, and the sound of each repeat. The number of repeats is normally governed on a delay device or plugin, by a control which is often labeled Feedback. The greater the feedback level, the more the number of repeats, which can sometimes extend into a never-ending cycle of feedback. Often located in very close proximity to the Feedback control are elements for dictating tone color. This can take the guise of a high and low cutoff filter or control, which allows the dictation of which frequencies are repeated, while excluding the frequencies you don't wish to hear.

The physics of how a delay or echo operates in the real world is all to do with soundwaves bouncing off surfaces. If you have ever stood in a large room with little in the way of furnishings or belongings placed on the wall, performing a single handclap will likely yield a form of echo or delay. The further you stand from the wall, the longer the echo or delay time will be.

Good delays gone bad

So this brings us to the point where echoes 'go bad' at which stage they turn into reverberation. To be fair to reverberation, or reverb as we tend to use it as a shortened term, it is entirely down to location and the number of reflective surfaces available. If you have ever walked into a large building or a long tunnel, you may well have been tempted to perform a clap, or if you're feeling particularly brave, sing a short note, purely to hear the reverberation. It is no surprise that our desire to do this is at its strongest when we walk into a building such as a large church or cathedral.

Being predominantly made of stone, the number of reflective surfaces, facing in many directions, creates an inviting environment for a soundwave to bounce from one surface to the next, in many different directions (see **Fig. 2**). It is under these circumstances that, beyond the beginning of our performed note, the resulting sound and tonal color blurs our initial sound source, decaying over time, as the reverb tail declines. The larger the building or structure, accompanied by surfaces that reflect sound waves, the longer

the tail will be. Consequently, it is not unusual that when trying this out in a cathedral, you'll hear a reverb tail extending for many seconds.

So to summarize, if you can hear your performed sound coming back at you as a straight repeat, you have an echo or delay on your hands. If the sound you're hearing is more of an audible mush, albeit a very pleasant sounding audible mush, you have a reverb.

Too many echoes

Where we get a small degree of crossover between the two effects is with an element known as pre-delay, which is found on reverb effects units and plugins. The purpose of this control is to induce a small delay between the initial sound and the start of reverberation. While this is always present in some minor capacity or other, we have been known to use this effect to our advantage, especially on instruments of a percussive nature such as drums. That effect, where you hear a snare drum with no reverberation for a few milliseconds, only to then hear the reverb appear, is where we have a longer or exaggerated pre-delay.

So if reverb is so cool and grandiose, why would we ever want to entertain using delay instead? The answer to that lies in the versatility of a delay. It can be used a little bit like reverb, but many producers prefer the versatility and control that you can get with delay, which tends to remove the mush-like status that we previously mentioned. We are not knocking reverbs – they're great, but control in production is a wonderful thing. Moreover, delays can add rhythmic color and cool effects to your own produced music. Let's explore how, in the coming pages.

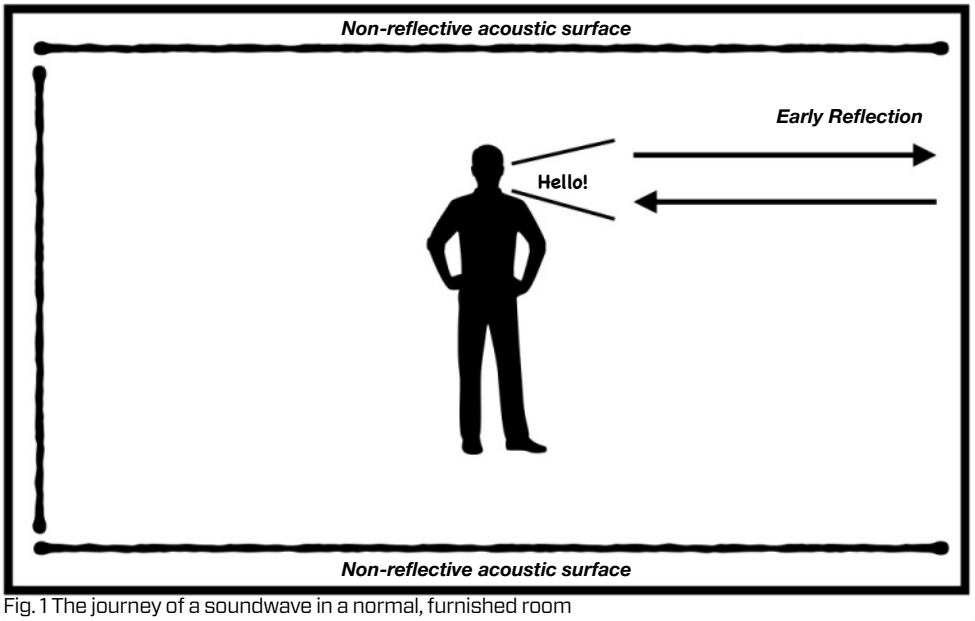


Fig. 1 The journey of a soundwave in a normal, furnished room

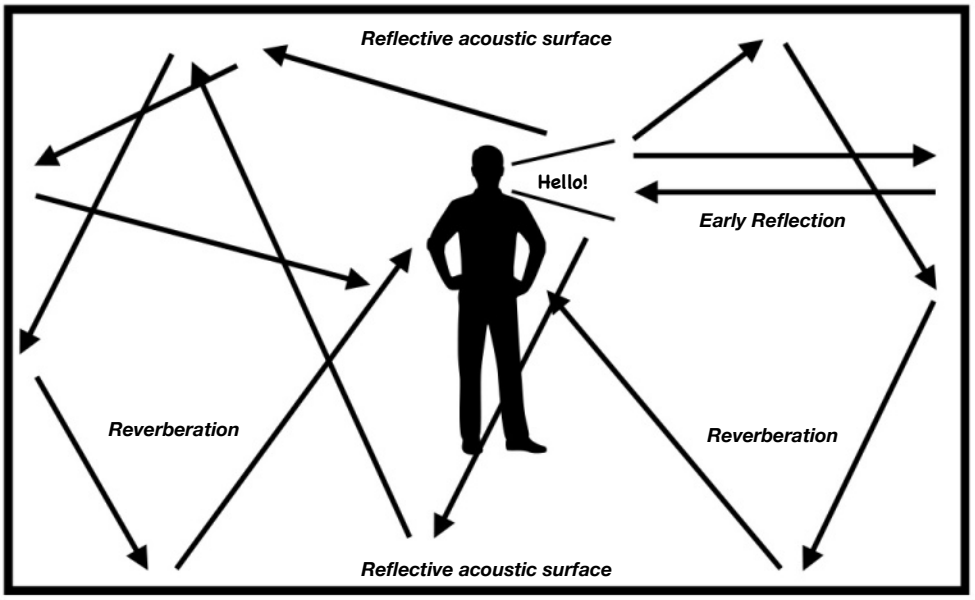


Fig. 2 ... Compared to its journey in, say, a stone mausoleum or spooky crypt

DELAY DECADES

All good effects begin with a great sound source, and they don't begin in a more organic setting than this...

Prior to the use of technology, if you wanted to create some form of delay in sound, you would literally go back to the basic principles of using reflective surfaces. With the advent of recording technology, specifically magnetic tape, it became possible to capture this effect using microphones, allowing the first produced music to be achieved, using effects. Obviously these were more organic than artificial, but the practice was sound enough.

As magnetic tape became more widely adopted, much musical exploration took place in the '50s classical music world. The style of music known as musique concrète, was heavily associated with the use of tape recorders. Also described as 'found sound', musique concrète relied heavily on what could be regarded as one of the earliest forms of sampling. Sounds would be recorded to tape, vari-speeded, re-recorded and then spliced.

This became an infamous process at institutions such as the BBC Radiophonic Workshop. Meanwhile, composers such as Karlheinz Stockhausen and Pierre Schaeffer were experimenting with the ability to record a sound source, such as an instrument, and then replay it moments later. This was possible due to the inclusion of a record head and playback head on a tape machine, which would allow the recording of sound, which could then be reproduced a few seconds later. Altering the speed of the tape machine could induce a change of tempo, and other bizarre effects, if the speed was changed in real time during performance. Here, we began to see the appearance of the tape loop, which has left its legacy in current production terminology.

Soon this primitive method of delay entered the world of commercial music. An early example is heard in rockabilly: a slapback echo, a very short delay time used on guitar tracks, substantially thickening the guitar's texture. This was popular with engineers like Sam Phillips from Sun Records, and early Elvis Presley.

Companies began to develop better solutions. The Copicat, invented by Charlie Watkins, and the Maestro Echoplex, both employed the same technology previously used in tape machines, but in a smaller and more bespoke package. Both machines relied on a loop of tape, fed through a record head, followed by a playback head. With the employment of more than one playback head, multiple delays could be created, while a speed control would allow the alteration of the delay or echo, allowing the creation of some very interesting speed and pitch based effects.

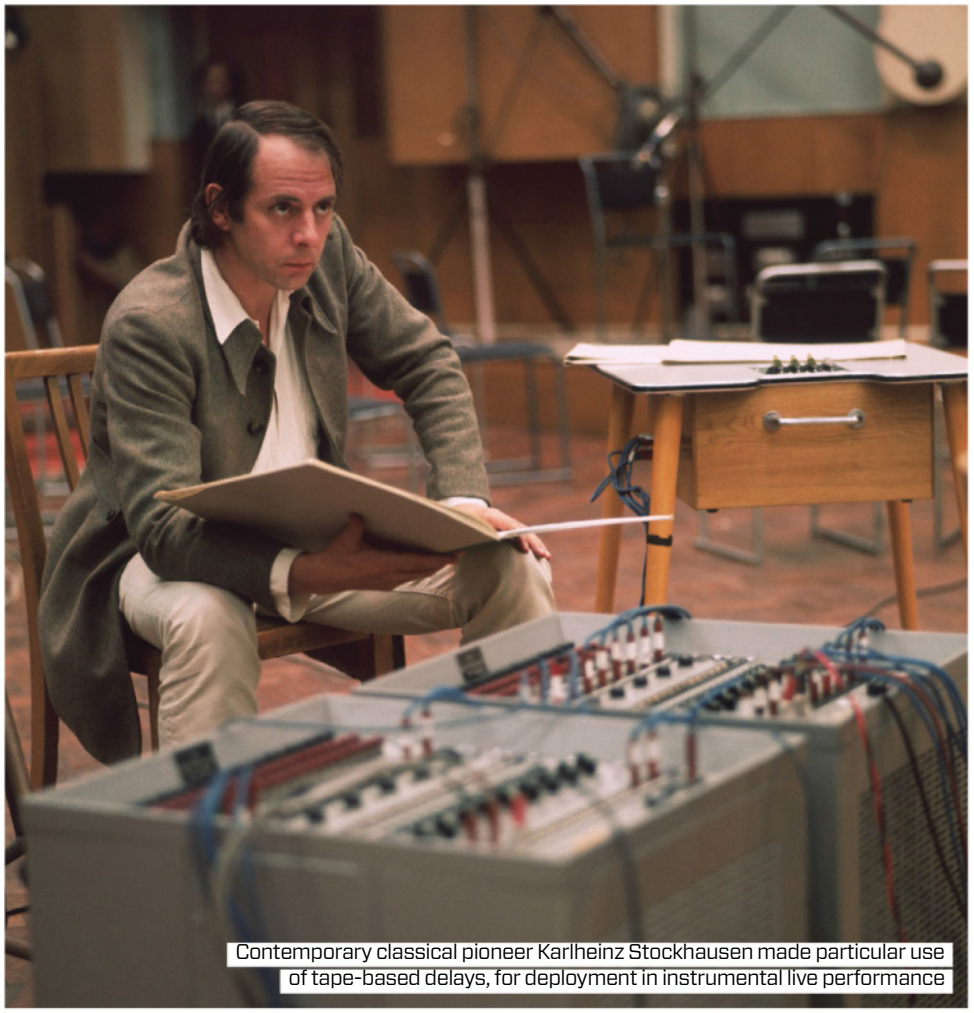
It wasn't until 1969 that one of the most synonymous terms, linked with delay circuits, was introduced alongside the invention of the

The WEM Copicat was one of the first convenient and portable tape delays

solid state delay circuit. Invented at the Philips research lab, the Bucket Brigade Delay (BBD) allowed the inputting of an analog signal at one end, with the outputting of a delayed signal on the back end. The circuit would allow for multiple delays to be applied, which would often decay along the timeline of the delay. By the mid '70s, compact BBD guitar pedals proliferated.

The inevitable progression of pedals and effects processes, resulted in digital pedals. The advantage being that longer delay times could be created, where there were distinct limitations for analog pedals, due to high noise levels emitted with longer delays.

The BBC Radiophonic Workshop were pioneers of tape usage, for both looping and delay purposes. Daphne Oram grapples with a tape loop



Contemporary classical pioneer Karlheinz Stockhausen made particular use of tape-based delays, for deployment in instrumental live performance

USING DELAYS

There is more choice in delays than there are repeats created from the feedback control, so we guide you through the best place to use certain styles and types

Delays are relatively uncomplicated, or at least that's the way it should appear, but there is equally plenty of diversity when it comes to delay types, and timbral color.

Let's start by highlighting some of the delay options, with an indication of where you might want to use them. Of course, there are no hard and fast rules with delays in the software domain, and you can use any delay, anywhere you like, freely. The key thing to remember is whether it does what you want it to do, and sounds as you would like it to sound. If it doesn't, you can be very sure that like delay repeats themselves, another delay will be just around the corner.

Going straight

Let's begin our tour by examining a basic, straightforward delay. Every DAW will be equipped with one of these; in fact it's quite likely that there will be various different delay styles. Many of the parameters are identical across all platforms, so we will begin by looking at the Echo plugin, which forms part of the core Logic Pro X package of tools.

There are three controls, which will dictate the basics for creating a successful delay. Other parameters will have an effect on elements such as tonal color, but we will start with the basics.

The first control will be labeled something along the lines of Delay Time. Unlike delays of yesteryear, where they were not synchronized to a computer through a DAW, the timing would be measured in milliseconds, albeit in a relatively crude and inaccurate way. As you can see overleaf, we don't have a timing control, but we do have a Note control. This takes the hard work out of adjusting your delay or echo, by snapping the delay immediately to the tempo of your track. Instead of working with milliseconds, we are working with note values, such as the 1/8 note value that we see in our example. The good news is, there are plenty of note values to choose from, such as 1/4 note and 1/16 note, with dotted and triplet note values too. There's more good news, because if you are unsure of the relative timing

implication of these note values, you only have to plug in your delay and try them out, to find out what suits your track's groove.

The next control in line will be a feedback option. The purpose of this element is to provide more and more delay content; the greater the feedback level, the longer the delay will take to dissipate. It is also incredibly common for the feedback control to be able to induce feedback, as the name would suggest. In this scenario, the delayed sonic element will repeat into a frenzy of overdrive, which was a particular effect that was popularized with some of the more extreme reggae/dub tracks produced in the late '60s and early '70s. It didn't take long for this technique to become fashionable with electronic artists too, and can be heard on tracks by the most diverse electronic experimenters, right the way through to pure pop by the likes of Madonna.

The final element is the blend between the wet and dry mix. In our pictorial example, we have a fader for the dry signal, which is the unaffected signal, and a fader for the wet signal, which is very firmly the effected signal. It is also incredibly common to see a single pot or fader on a delay plugin, which will see-saw or blend between the two signals. The overall result is the same; it is merely the process by which we get there.

Armed with these basic elements, it is highly likely that you will be able to navigate 99.9% of all delay-style plugins, but bear in mind that this is just the starting point. There will almost certainly be additional parameters, for controlling other elements. In the case of our featured Echo plugin, we also have a Color fader. This parameter controls the frequency band that is being selected for decay. Moving the fader to the far right will induce a brighter repetition, while the left will concentrate on the lower frequency band. Again, it is very likely that you will see features such as this on other delays, particularly tape delays, and it might take the guise of two faders or pots, allowing you to assign a high frequency cutoff point, and a low frequency cutoff point.

Doubled up

Despite what might be viewed as the overly obvious side of delays, they actually have several tricks up their sleeve. One such trick can be applied to create a thickening of texture, and is ideal when working with vocals, particularly if your recorded vocal track feels a little thin. If we apply a single delay to a vocal track, but keep our delay time short and feedback amount low, we can make it sound as though our vocalist has been double-tracked. This will have the effect of thickening the texture of the vocal, without the necessity to record the second pass.

When working on this you will need to have a delay that displays delay time in milliseconds. Add your delay, and ensure that you're only hearing a single repeat, which you can normally achieve by keeping your feedback level exceptionally low, but then ensure that your delay time is set between the 25 and 40ms timeframe. Anything longer than this and it will just sound like a regular delay. Anything shorter, and you're likely to start hearing flange-style effects, which may or may not be desirable, depending on the effect that you're trying to achieve.

The Haas effect

While our previous doubling technique is effective enough, we can take this concept to the next level by using an effect which was named after the German mathematician, Helmut Haas. This technique delays our original signal, much as before, but in a way that creates an enormously wide sound-stage, for the applied signal.

Firstly, ensure that you have a recorded vocal part in mono, which you can pan hard left (see below). Next, apply an Aux or Buss to your vocal part, and send the same amount of signal to your auxiliary buss. Now plug in a delay; we only want a single repetition, preferably without coloration, as we want to effectively achieve a carbon copy of the original, on our Aux or Buss. We are using Sample Delay, from within Logic, which provides the ability to view your delay as a millisecond timing, while providing a single repeat, which is identical to the original signal source. Pan the effect channel hard right, and increase the delay time while listening to your track. As you approach the timing of around 25ms, you should hear an extraordinary thickening, which makes the vocal sound incredibly wide and impressive. The exact timing you set for your delay can head up to around 40ms. It does depend on the music you've recorded, which is why it's so important to listen as you apply the delay. When it feels right, leave it, but you can fully expect the timing to be between 25 and 40ms.

While we have discussed the use of this effect style using vocals, it can be applied to numerous other instruments. Some will provide better results than others, but you can get some very

effective results using the same technique on drums or percussion.

Getting longer

As we increase our delay times, we begin to explore other named styles of delay.

The next one in line is described as slapback echo. A throwback to the good ol' days of rockabilly, it's particularly effective when applied to guitars, and even upright basses, playing in the traditional rockabilly style. It's a technique that was very popular with a number of guitarists, including British instrumental band The Shadows, and even Johnny Cash.

It's a stylized delay, and it might not be suitable for all purposes, but if you do find yourself using it, you're likely to be increasing your delay time to anything up to around 250ms.

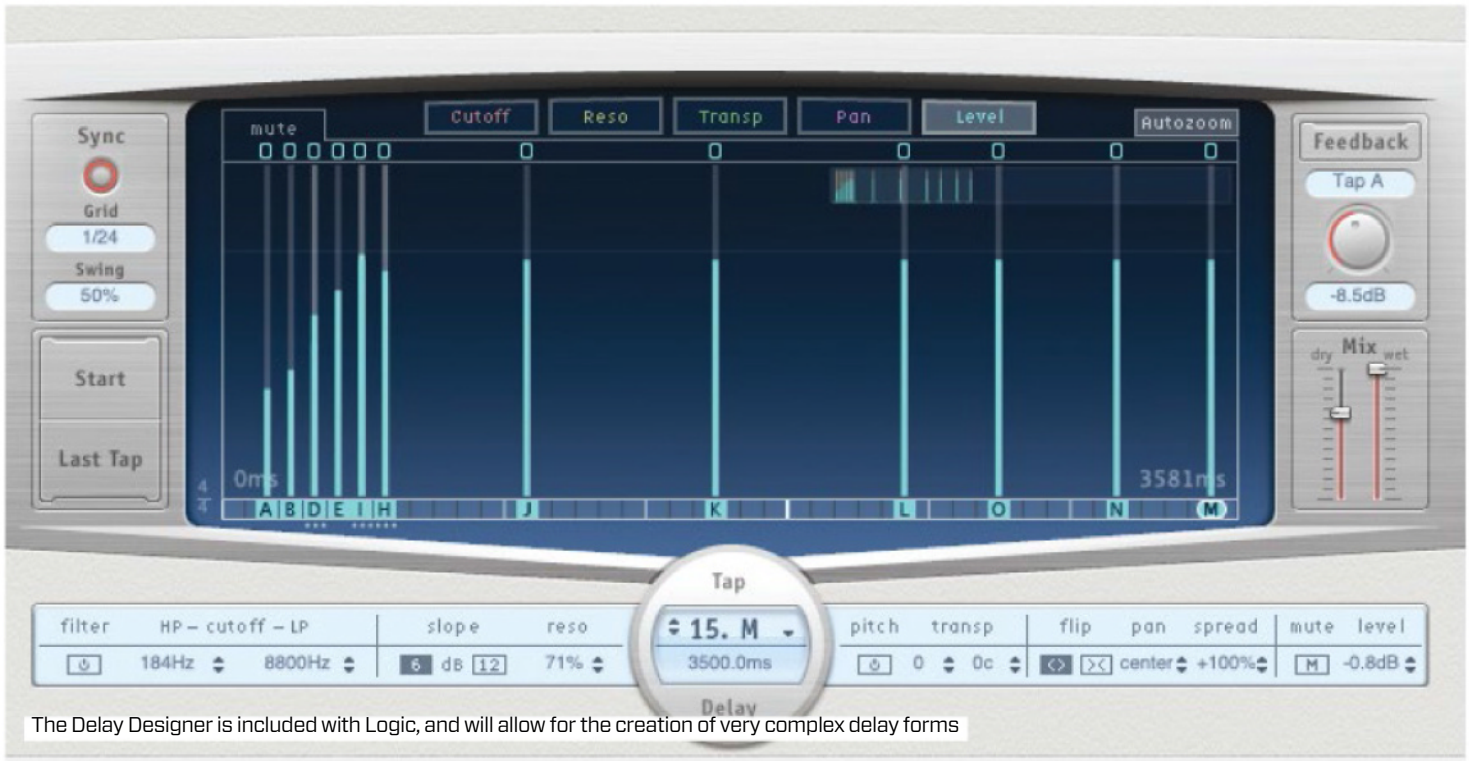
In the context of a delay which is synchronized to your DAW, look to apply a note value of 32nd or 16th note, which should yield something along similar lines.

Going stereo

With the march of technology, delays moved from a mono signal to the stereo domain, predominantly with the advent of digital delays. This opened up a huge world of production, where delays could correlate in stereo. Of course, nowhere is this more obvious than within the software domain, which we will come back to shortly.

But what would you do if you wanted to create a stereo effect from a mono signal? We only have to look at an instrument such as the Fender

The EchoBoy delay plugin from Soundtoys and its legendary ping-pong delay, definable on both left and right channels



The Delay Designer is included with Logic, and will allow for the creation of very complex delay forms

Rhodes electric piano to see how this might work. The model which was described as The Suitcase was equipped with a large amplifier and speakers, that would sit below the electric piano itself.

These models were also equipped with a form of tremolo, that would literally ping-pong the Rhodes signal, from one side of the stereo spectrum to the other. Equipped with a speed control, you could select how fast you wanted this tremolo to be. While it could be considered something of a crude device, it was incredibly effective at increasing the width of the sound stage for the instrument.

While this example is very much not a delay, the concept of the ping-pong delay is largely the same. An inputted signal source would then reappear as a delay on either the left or the right-hand side of the stereo spectrum. This form of ping-pong effect is often obtainable with delays within DAWs, which are merely described as stereo delays, while some delays may have a specific mode, which will allow this form of operation. One such example comes from Soundtoys, whose EchoBoy plugin (see left) has a specific ping-pong mode, where it's also possible to change the timing for the left and right channels individually. It is also important to note that you'll want to increase the feedback control, by some degree, to facilitate the sustained element of this effect, which is a unique attribute for the ping-pong delay.

If none of your current delay plugins seem to cater for the ping-pong delay, try searching out a stereo delay from within your DAW. Apart from the possibility of recreating some form of ping-pong delay, the chances are that you'll also find plenty of other stereo delay effects to use, along the way.

Future taps

As the technology of delays has improved over time, so more complex delays have become available, sometimes built in to synthesizers or other effect types. Many of these more elaborate delays take on the guise of what we would refer to as a Multitap Delay. The clue here is very firmly in the name, as delays of this kind can often have the capacity for a greater number of repeats, coupled with good control.

Some delays have the capacity to literally 'draw in' where you may like a delay, as well as the volume level that you'd like to have applied to each delayed repetition. You may well have something like this in your DAW already, but if not there are plenty of further options. By way of example, the Delay Designer to be found in Logic (see left) allows the user to draw in delay repeat points, accompanied by their volume amounts, which you can see in the example from the height of each bar. Furthermore, complex delays such as this may also have the capacity to alter pitch, alongside the volume.

Eventide Ultratap: a very contemporary delay, giving instant delay-inspired interest, for any commercial production

If all that feels like hard work, fear not, because these forms of delay are often accompanied by numerous presets to try and get you underway.

Another fine example of a Multitap delay, comes from Eventide (see above), with their Ultratap plugin. It lacks the visual interface, that we see on the delay designer, but it gives us numerous controls for contouring the behavior of each delay, dictating timing, volume and modulation of pitch. Again, it is stacked full of presets, and a real 21st-century delay solution.

Get creative

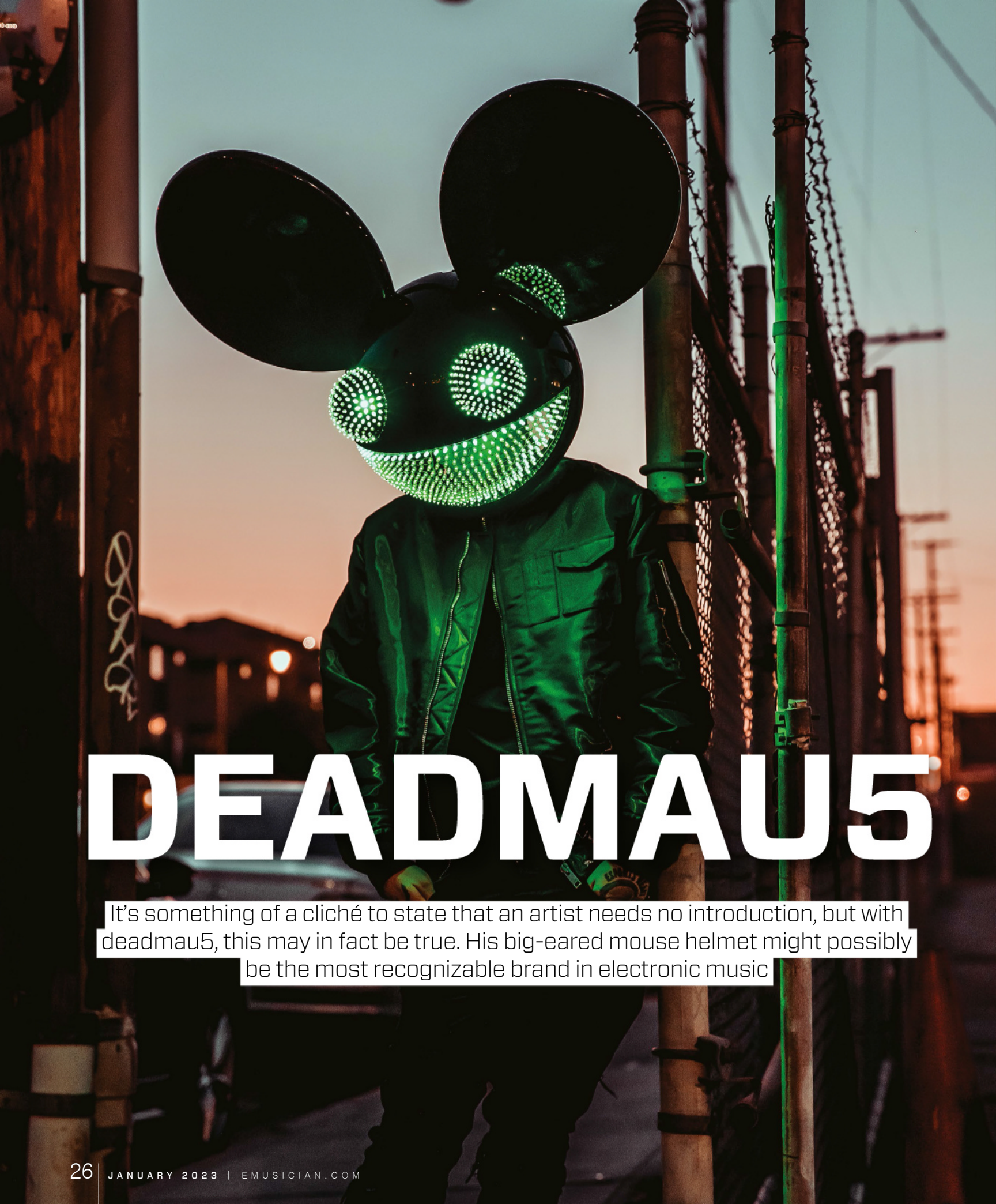
It's so incredibly easy in the 21st century to be somewhat lazy with a delay, just plugging it in to your instrument or audio channel, and blending it with the initial signal. Through the use of auxiliaries and busses, within your DAW, you can of course take this process much further, by allowing automation and control of the delay element itself.

Here lies a form of delay-based nirvana, which is ripe for creative exploration, and who wouldn't want to create a unique and individual sound, which can often only be done using software? Look at an artist like The Edge, from U2; his guitar sound is so reliant on delays, it's difficult to imagine him without it.

Saturation in software

With the move away from analog technologies, and specifically tape delays, we arguably left something behind when we went digital. That something is the sound of magnetic tape. Apart from the physical noise of the tape whirring within the machine itself, tape brings a degree of distortion and saturation to proceedings, which we generally tend to like as a sound color. Thankfully, software companies appreciate this, and many tape delay-style plugins are available, either as an included component within your DAW, or as a third-party freebie or purchase. The chances are, any delay which makes use of the word 'tape' in its name or description, will make use of some form of saturation or distortion effect. The aforementioned Echoboy, from Soundtoys, is a fine example, as it is equipped with a number of saturation colors, mirrored from vintage machines.

Of course, you do not have to make use of this element, but we think that once you've found it, you'll probably want to use it, because it sounds so damn good.



DEADMAU5

It's something of a cliché to state that an artist needs no introduction, but with deadmau5, this may in fact be true. His big-eared mouse helmet might possibly be the most recognizable brand in electronic music

Finding globe-sweeping success in the late '00s, deadmau5 played a pivotal role in pushing electronic sounds into the US mainstream, becoming one of the world's highest-played and highest-paid electronic artists in the process. Grammy nominations, platinum records and world tours eventually followed, as the producer courted planet-wide adoration and controversy in equal measure over the next decade.

Fast forward to 2022, and deadmau5 remains as influential as ever, and no less exploratory: continuing to produce electrifying singles that reimagine progressive house with each release, he's nurtured the careers of like-minded artists through his label, mau5trap, and entered into the software world with the development of OSC/PILOT, an innovative performance tool for electronic artists.

For a man who's notoriously averse to interviews, we find mouse-headed super-producer deadmau5 in a relatively good mood. Speaking to us, sans headgear and cigarette in hand, from his home in Ontario, Joel Zimmerman seemed content to spend an expletive-sprinkled time talking shop, painting a picture of an artist equally excited by the technology used to create and perform music as the music itself.

Could you tell us about some of your favorite software tools?
I don't know, man. I guess everyone was like 'fuck it' during the pandemic. Nothing really crazy has really advanced that far. To be honest, I feel like we've kind of ceiling-ed on music software. There's a lot more of the same that's come out, like... 'oh hey, it's a new reverb!' Nothing's really stood out. But then again, I haven't really been looking.

So you think we've hit a wall in terms of advancements in music software?
Well, you can always do better, faster, more efficient. But in the realm of DSP, I almost feel like we've hit it all. It's just a lot of the same. And I'm just like, waiting for a lot of things to get overhauled and done a different way. I'm seeing some new DAWs come out, and stuff like that. The ones that I'm liking the most are the ones that have a lot of really cool protocol control inside of them.

Ableton's been cool, just because it's 18 years in development, it's nice and stable and all that stuff. And it's shell-ready. But it really lacks in protocol modularity. It's just like, 'oh, here's MIDI. Here's an antiquated 60-year-old protocol you guys can still use'... like what? Just have OSC. And they do, but you have to do it in Python or use Max for Live, and there's no native OSC stuff. So actually, a lot of the time I've spent during the pandemic was developing a VST that will allow for the interchange of different protocols in and out of any DAW.

What DAW do you find you're using the most these days?
I switch between all sorts of things depending on what it is that I'm trying to do. For MIDI-deep stuff, I tend to go to Cubase for better control. Because Ableton is not so great for that, it's good for quick and dirty. So if I just need something quick and dirty, I'll just go in there and do some quick edits. It just depends on what it's called for. Because a lot of the mixing I'll do on my desk, and neither Cubase or Ableton supports SwiftMix for my Neve console, but Pro Tools does. So I gotta use that for that, so I can recall faders and stuff like that on my desk. It's a big mess of different DAWs for different things.

> "It's not like I had a friend within 100 miles of me who even knew what a tracker was, much less make music on a computer alone"

We wanted to dive into the production behind your latest single, 'XYZ'. What was the approach behind this one?
I don't think I really had one. It was just like, 'okay, time to sit down in the studio and crank one out.' It was just a lot of farting around and tooling and playing with some synths, and just coming up with this main hook thing. And I was like, 'okay, I can work with that'. And then I just built around that.

Were there any particular synths or effects that you used that were really fundamental to the track?
I did get to finally try out VCV Rack 2.0. I've been waiting so long for this. VCV Rack has always been great. But 2.0 finally has sample-accurate sync with the plugin in a DAW now. The other one, you had to use a host and it was a little late here and there, and the sync wasn't nice and all of that stuff. But now in VCV Rack 2.0 Pro, they have a host that will work in any DAW that's sample-accurate, finally, and it's fucking amazing.

Because a lot of the modules, these days at least, with modular racks, if you go out and buy a bunch of mods, chances are one in three, it's going to be a digital module, which means it's just lines of code. So there's all the Mutable Instruments stuff, and all the really cool fancy expensive ones, you can buy for like next to nothing through VCV Rack, and they're the exact same module. Even better, because you don't have to record the analog path. It's just printing it digitally, right there.

A lot of those oscillators are great. Obviously, not so much the analog oscillators like you'd get out of a Cynthia or something like that. But the digital oscillators, all the wavetable stuff and the digital FM ops, and stuff like that are great. So I actually did get to use that on 'XYZ'. That was kind of cool.

Do you have a physical modular rig too?
Oh, you clearly didn't Google that. Yeah, I've got a 30-foot wall. I've been known to collect the analog modulars.

What was the last module you bought?
The last module I bought was probably my third Yarns from Mutable Instruments, because like I said, it's got to be at least a good 20, 30 feet. Having just one MIDI-to-CV converter somewhere in there is kind of inconvenient, because I have to patch out to mults, and then into other things. So I had to get three to space them out throughout the system, so I can reach cables to certain banks of oscillators without having to go through like 20 mults just to get to the other side of the synth.

Having a modular synth that large, is there ever a point where you're confronted with too much choice, too many options?
Yeah, but it's not crippling. It just means I get a different result every time, which is kind of nice, as opposed to just doing the same thing over and over again.

Looking back now to when you first started producing, you were using Impulse Tracker in the early days, right?
Yeah, actually. Wow, see – he knew that. [*laughs*] Actually, if you want to be stupid

about it, my first song was written in Turing. Then I upgraded to MS-DOS. Jeffrey Lim, the Australian guy, wrote Impulse Tracker. Scream Tracker was their other one, and those were the two big trackers back in the early '90s. I did cracktros – that was fun.

So how did things progress from there?
I hung around in that scene for a while, and then met some dudes and hung out with like-minded people who did the same thing, and then showed them what I was using, versus what they were using. It was a very small collective back in the day, because the internet was not as big. And there definitely weren't any real life clubs of people using trackers or anything like that. It's not like I had a friend within 100 miles of me that even knew what a tracker was, much less knew they could make music on a computer alone. It was just a case of putting feelers out there, putting in some footwork and meeting a couple of like-minded people. Then moving within those tribes and creating opportunities for myself to apply those skill sets into real world practicalities, like working in a radio station, or making advertisements or composing music for stock library stuff. And that's kind of where it all stemmed from.

Do you still use trackers today?
Not really. As a matter of fact, I don't even think I could probably get around an Impulse Tracker any more. I've seen it, I know what it looks like in my brain. You knofor you koprijohaerghog aerhou['garewhjiogaerhiopgehrhijpw the old monitors when they would burn the image into the monitor? That's what I'm seeing in my brain, but I don't think I could figure it out. I don't know if it's like riding a bike. They're all very programmatic. It was a lot of using the arrow keys. And no mouse – once you load the tracker, you just put the mouse to the side and use your keyboard and navigate around that way.

There are some plugins that will emulate what trackers do. Even then, if you just want to run a pure tracker, you've got to run a virtual machine or something to emulate that MS-DOS functionality and all that stuff, too. I think that's all deprecated in Windows now.

Would you describe yourself as more of a keyboard-and-mouse producer or someone who's more tactile, and likes working with real instruments?
Keyboard and mouse all the way. Some people will have that kind of muscle memory and those skill sets ingrained in their brain to be able to put their hands on some tactile instrument and put out their idea and do whatever they want. But I've gone through those motions where I tried to train myself to be a proficient player, but the right brain turned on and I'm like, well – why? So I just kept doing what I was comfortable with. There's no better way to do that, and everyone's different. For me, it just worked out that I've just got a very input/output brain.

How often do you find yourself experimenting with new gear?

> "I've gone through those motions of training myself to be a proficient player, but the right brain turned on and I'm like, 'Well – why?'"

I've got my mainstays, which contribute to the tone and sound you'll hear in my music. You know – you hear a fucking saxophone and you know it's Kenny G.

I do introduce new stuff from time to time just for fun, and playing with it and experimenting and stuff like that. But I'm not like Richard Devine, where I get a new gadget every 20 minutes. He probably just got another one in the middle of this call, and then he's rocking away on it, but that's him. You know what I mean? He's someone who loves doing that shit. I love that guy.

I just want to know where he puts it all. There's no way he's fitting all that in the studio. This guy's got to have a warehouse full of shit. So many companies are just like, yo! Try this. Try this. Try this. If you follow him on Instagram, it's great. If I built a synth, I'd get him one – for sure.

When you say mainstays, could you go into more detail on what instruments you're talking about?
I really, really love my Colossus these days, the Analogue Solutions Colossus, which is a kind of an EMS Synthi 100 clone. Which was like a big old archaic synth, and nearly impossible to get in good condition. This guy Andrew at Analogue Solutions makes these clones of it. Obviously they're huge, they're massive, they take up the whole fucking island in my studio. It's got 16 oscillators, you can't go wrong there. Of course I'm only really using like four or five in most cases, I have yet to max the thing out, but it's a really good canvas for me to start creating simple patches and stuff like that.

Also the Moog Voyager, I can't seem to escape that synth. It's the last good Moog. Everything else has just been kind of... don't get me wrong, bless Moog, but they cheaped out after that.

When it comes to drums, do you synthesize your own or are you working with any sample libraries?
It's a mix of both. Sample libraries are cool, but sometimes you want something really bespoke. So it's just digitally compositing some synth sounds with some transient sounds from here and there. I don't spend too much time on it, but from time to time, I'll make my own little samples to use. But more often than not, they're just straight recordings of synth lines and stuff like that from the gear on the wall.

Is there anything on your wishlist studio-wise?
I want the Super 6. I can easily get one, I just haven't yet. It's simple, but I hear it sounds great. I like that kind of stuff – no bullshit, no surfing through a tiny LED screen. Nord Leads are cool, they have a sound, but it's just getting around those fucking things, like come on. That's also why I like the Voyager. It's all just right there. Just show me where the knob is and that's it. I don't wanna go menu-diving.

There's definitely something to that instantaneous quality where you can get going straight away.
Yeah, which is great. But in a live setting, not so much. Because then you gotta remember what was what. At least for the Voyager it's kind of 50/50, where you can kind of program a patch and then recall it with a PGM change or something like that. But with something like a Model D like, you know – good luck. [*laughs*] You're gonna get a different result every time – which may be what you want – but not close enough to the original.

I've read that you try to master your tracks as you go along, rather than leaving that process until the end. Is that still an approach you take now?
Sure, but there's obviously some time at the end where I'm like, I'm done, quote unquote. Then I'll spend a little more time cleaning up the arrangement and the bells and whistles and all that crap, and listening through and doing my thing as expected. But I do tend to get the middle part of the song about 80% of where I want it to be on the release before I've even finished the song. It's just how I like to work, because there's something more motivating about working on something that sounds good, versus something that sounds shit, where you're like – oh, it'll sound good when it's done.

I think that's what deters people from music in general. A violinist will be playing by themselves and thinking it doesn't sound great – but it will when there's four of us. It's

> "I wish I had a way of explaining without using that word. NFT has this connotation that makes me cringe. And I'm involved in it"

distracting, and some people can't envision those extra layers. It's kind of the same brain when it comes to making electronic music.

I think more and more these days, people want instant results.
Oh fuck, yes. Absolutely. It's just getting worse and worse as we move through time. Or I'm getting older.

What do you make of these stripped-back, one-knob plugins that keep coming out?
It's cool, it's forward-thinking, and I like that. But that opens the door to homogeneous production pipelines.

I guess there's a good side and a bad side.
Yeah, it's fucked up. It's like the whole CG community that has infiltrated electronic music, or DJ culture, or festival culture. It's absolutely a joke. The least-paid intern at Method or Blur Studios could run circles around some of these guys. A guy shows up one day to the very first music festival and says, 'I'm a graphics guy!' – and then he busts out a little laptop running Resolume and then all of a sudden he's commandeering 16k worth of visuals at EDC.

Then you put them in a professional setting, with people who have actually been doing this their whole lives, and he's a fucking joke. But he had those tools, you know what I mean? It's spawned this huge industry of fucking shitbags. Okay, not shitbags, but just... incompetent, or people not quite at that level yet. Or people who are at that level because of this software. It's like a raccoon population in downtown Toronto, that's the best way to fucking see it.

Sometimes there's no escaping that. And then the bar is just so fucking low, and the same could be said for music. Someone's got that cool idea, and then applies that one thing that no one bothered to do, because no one's thought of it. There's always got to be a first. Then that'll take off and then someone will emulate that, and then someone will find out how everyone's been emulating it. And then it'll just become a Web3 application where you can go and make fucking EDM bangers.

You're involved in the Web3 space a little, right? Could you elaborate on that?
It's like Web2, but one better. [*laughs*]

Do you have an NFT project cooking?
I wish I had a different way of explaining that without using that fucking word. NFT has this connotation that makes me cringe. And I'm involved in it, you know what I mean?

What excites you about the technology?

The way that blockchain can be used as a store of information or value to a holder. That's the basic breakdown of it. I know, it's like... 'but the monkey's actually mine!' That's not what that means. It's a different layer of abstraction of where an artist can exist in a space, and have the tools and protocols to give them more control over what they do, versus the traditional method of 'oh, I'm starting to make music, shit I need a manager to manage me, I need an agent to agent me, and I have to get on to Apple and Spotify to be heard, I got to make a TikTok so little idiots can watch me do dances'.

There's real traditional methods to get music out. Or maybe not get it out, because no one ever got famous because they're really huge on the blockchain. But I mean, the monetary stuff, the control stuff. Because the end of the day, okay, you did all this work and 10 years into your fucking career, but you're out conservatively 50% of control over your own shit. Because you got your label, you got your manager, you got your rights, you have neighboring rights and all this fucking shit too. The blockchain lets you govern more of that, if not all of it.

So the appeal for you is the potential to build new models for publishing and royalties?
That's the idea, but it's been an uphill battle, no thanks to this fucking NFT shitshow. That's really put a bad stigma cloud around it. Where's 3LAU? Where did you go? What happened with this big fucking steam? Oh, you spent it all in a week. That had some really good fucking steam behind it. But what happened? I'm not saying it's his fault by a long shot, he did a great thing. But the whole stigma of the coin ownership or the token ownership and all that got muddled in with some fuckhead in New York selling a fucking picture of a monkey and pissing everyone off.

That was a big step back for DAOs – even though they have nothing to do with each other. You start throwing around acronyms at people and then somewhere along the line, you're gonna have to say non-fungible token.

I suppose in the mind of the average person those two things may get conflated.
Oh, everything gets conflated. It's like explaining critical race theory to somebody south of Virginia. They're like, 'oh, I hate it!' – and it's like, do you know what it is? 'No, but I don't like it!' It's just stupid. But it is what it is. All we can do is push forward with the technology, if it's proven, and do what we do.

What would you like to see developed in the future in terms of music technology?
I enjoy a good surprise. But I would love to see more protocol support in the DAWs, up to and including DMX. Or scripting capabilities that are native to the DAW and not some bullshit workaround that has to make it run a different API on a different thread. Because Max for Live does not run natively in Live, it's not part of that same thread. It's just another executable inside it. It's no different than running it side-by-side on the same machine.

I'd like to see more native protocols in applications. There's a new DAW called Ossia, it's fucking crazy. But it's procedural. If you're familiar with DAWs, and you open that up expecting to make something in the first hour, forget it. If you're a CGI person reading this, it's like the difference between 3DS Max and Houdini. It's all very procedural, network-driven, like node topology, shit like that – it definitely is not for everyone. But Ossia is a protocol heaven. It'll do OSC, it'll do DMX. And it's all native into the engine too. It's really cool. But it'd require so much tooling for me to be able to produce a massive live project that will be performable and entertaining for up to two hours. ■

'XYZ' is out now, and will appear on the* We Are Friends (Vol 11) *compilation due out later this year on mau5trap. deadmau5 is on tour now as part of the deadmau5 Presents We Are Friends tour .

Plugins 'n' stuff

Can you tell us about the live performance software you've developed, OSC Pilot?

We're kind of always tooling at it. There hasn't been a major update lately other than, we got it working and compiled on Metal, which means we can have it on iOS and iPad, which is really cool. My favorite thing about OSC Pilot is that we really lit a fire under Hexler's ass. He wrote Touch OSC 2 to keep up, and he really did a good job with that too, so kudos to him. We're both developing the same things in parallel.

There were a lot of things lacking in the PC department, or the multi-platform department that allowed me to use what is to me a great protocol for music communications, where it's note data, float data, controller values and stuff like that. I was like, 'how has this not replaced MIDI'? And where is this MIDI 2.0 that has warranted a whole coalition or group of experts that have just been talking about the spec for years and years?

It's like, fuck you guys. I'm not waiting on this. I'm gonna use an existing protocol that is open source, and I've developed my own tools around that. And all the better if people can find it useful, I'll put them into a more usable application for everyone else. So it's not like I'm pouring my life savings into this application because it's going to be huge. It's a very fucking niche group of people, but as long as I'm one of them, then it serves me a purpose. And that's great.

Stevie Wonder — 5 Sides Of A Master

Isn't he wonderful? The musician who overcame disability to become one of the greatest songwriters of his generation provides no end of inspiration to this day...

By Jerry Kovarsky

It seems like we've always had Stevie Wonder as part of our musical landscape. He started off as "Little" Stevie Wonder back in 1961, and had his first hit two years later, at the age of 13. Since that time he has grown and developed into one of the most accomplished and musically diverse pop artists of his generation. Popularity and success aside, let's celebrate some of his musical skills, of which there are many.

A Harmonica Virtuoso

We can see from his very first hit record ("Fingertips") that Stevie could play up a storm on the chromatic harmonica. In his earlier years he often featured a harmonica solo in his recordings, like 1968's For Once In My Life. Here he is playing over changes like a jazz player, with a deep, full sound, even at the age of eighteen. That same year he had a hit single with Burt Bacharach's tune "Alfie", and his live performance from 2012 honoring the composer is just stunning. Watch it at **bit.ly/EMstw1**

A Clav Master

Whenever musicians discuss playing funk on a Clavinet, Stevie's seminal track "Superstition" is referenced. But there are plenty of other tunes to enjoy his funk clav work on. His album *Music Of My Mind* (1972) featured a lesser-known track called "Keep On Running." Its percolating and driving clav riff is funk perfection. Along with "Superstition", Talking Book also features the mid-tempo tune "Maybe Your Baby", which has a less-busy wah-wah clav part that carries the tune beautifully. Innervisions (1973) has the monster clav track "Higher Ground", another multi-tracked clav opus that has Stevie playing the main riff, mimicking a rhythm guitar, and adding clav fills and melodic filigrees to cover the whole

> "Starting with his 1972 album *Music Of My Mind*, Stevie started playing almost all of the instruments on his recordings, carrying this through his next four albums"

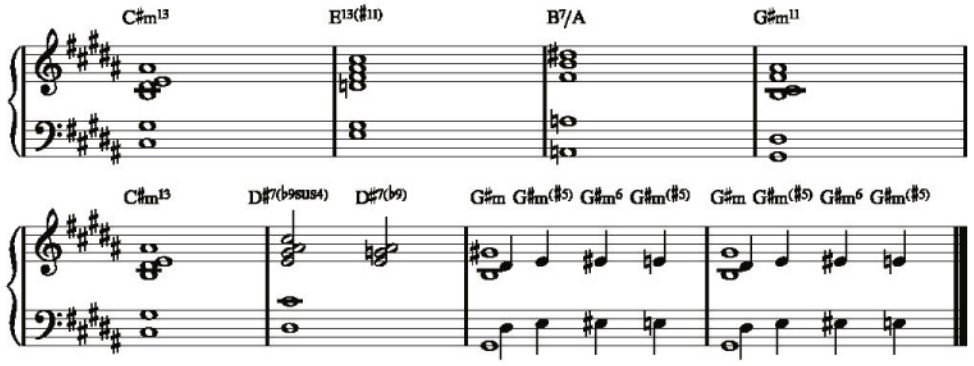



Fig 1. The harmonies for the intro to "Rocket Love" showcase a surprisingly advanced jazz influence.

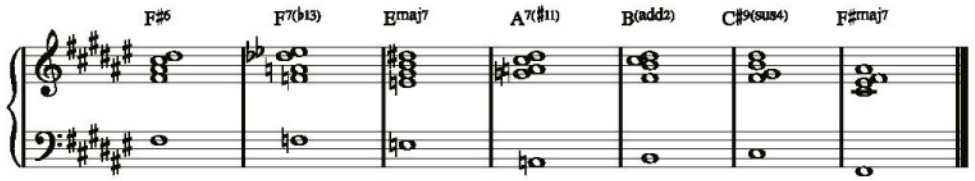


Fig 2. The chord voicings used in "Summer Soft" utilize the technique of retaining the top note of the chord while changing harmonies beneath it.

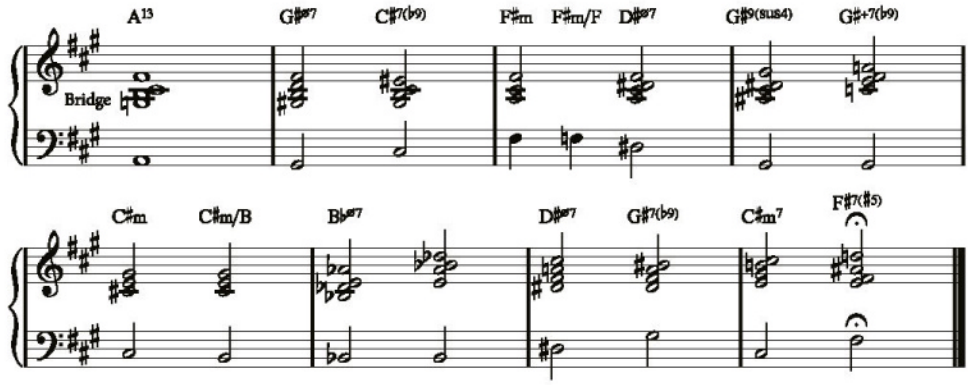



Fig 3. The bridge to Stevie's classic ballad "Visions" is a lesson in beautiful harmony and excellent voice-leading.

track, supported only by bass and drums. One year later, his next album Fulfillingness' First Finale features the deeply funky "You Haven't Done Nothin'". Take notice of how varied his touch and articulation is on the keys — in his hands it is so expressive. **bit.ly/EMstw2**

Badass Bass Player

Starting with Music Of My Mind, Stevie started playing almost all the instruments on his recordings, carrying this through his next four albums. He would play bass on one of his synths: early on it was the ARP 2600, then on the legendary TONTO massive modular synth system, and later moving into the Minimoog. Some tunes only required simple nondescript parts, but others were a lesson in virtuoso synth bass playing. The pinnacle has to be the amazing part he created for his 1974 hit "Boogie On Reggae Woman". Rythmically busy, with slides, bends and great articulation and feel, it is a tour de force performance. The bass part he came up with for "Too High", from Innervisions (1973) is a stand-out part of the song. And the bass sound and playing on "So What The Fuss" is the definition of Phat. **bit.ly/EMstw3**

Deep Harmonic Chops

When we think of hit pop tunes and commercial music in general we don't expect to encounter sophisticated harmony. But Stevie absorbed so much music in his developing years, and as his career progressed he often slipped in advanced and beautiful harmonies. A great example is the intro to "Rocket Love", from his 1980 release Hotter Than July. Here's the chords (**Fig. 1**), voiced a little jazzier than he does on the record. **bit.ly/EMstw4**

Another example comes from the tune "Summer Soft", from Songs In The Key Of Life. The chords in the verse keep the same upper note, and vary the harmony below it in a very sophisticated fashion (**Fig. 2**). **bit.ly/EMstw5** Then, the bridge of his 1974 tune "Visions" sounds like a cross between Chopin and Jobim, with such beautiful and logical harmony (**Fig. 3**). **bit.ly/EMstw6**

Give The Drummer Some

From an early age, Stevie would often take to the drums in live shows, delivering an energy-filled performance; like this one from 1971. **bit.ly/EMstw7** During his golden years he is most often the drummer — that's right, he opens "Superstition" with a groove he came up with while jamming with Jeff Beck. A concert in Japan in 1982 has him laying down a solid groove, and this clip has him dueling with master drummer Gerry Brown in 1995, and Stevie kills it. **bit.ly/EMstw8**

Ever Exploring

Stevie has always made use of new technologies as they became available. From his early adoption of the Clavinet Model C in 1968, he has always sought out new keyboards and recording technology. I've already mentioned his use of analog synths, but he was also one of the first to use sampling, on his Secret Life Of Plants project in 1979. It is well known that he was the impetus for Ray Kurzweil to develop the K250 keyboard system, and Wonder was always buying and using whatever latest development came out of the keyboard industry.

It's not only keyboards that Stevie liked to explore. This great clip from 1972 showcases him running his ARP 2600 through a talk-box, **bit.ly/EMstw9** and in our next clip he demonstrates it for a less funky effect. **bit.ly/EMstw10** In recent years he has adopted a very cool stringed instrument called the Harpejji, and he enthuses about it at this event. **bit.ly/EMstw11** He gave it worldwide exposure when he performed in tribute to Bill Withers at the Rock & Roll Hall of Fame in 2015. **bit.ly/EMstw12** ■

EASY GUIDE

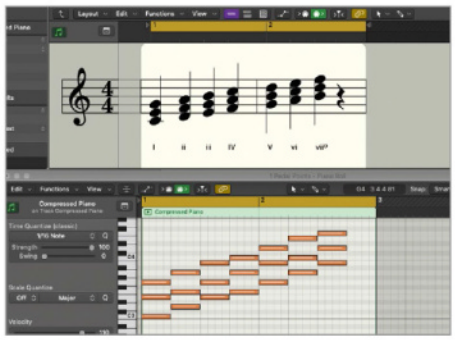
Pedal points

Our theory maven puts his foot down and explains why sticking to one note can instantly power up your progressions

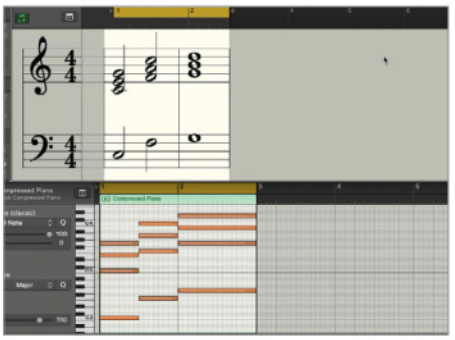
A pedal point – AKA a pedaled note – is best defined as a sustained note that continues unchanged over a set of chord changes. In classical organ music, these kinds of sustained notes in the bass register were played by foot on the pedalboard. From a sustained single-string note floating over a breakdown in a club tune to a single pitch ringing out in a series of guitar chords, pedal points are everywhere in music – but why?

The chief reason for them is to increase tension or excitement in a chord progression – broadly, excitement is the perceived mood with major keys, while tension tends to be the feeling that comes through in minor keys. Imagine Faithless's 'Insomnia' without that insidious, unvarying sequenced B bass note through the first half, and the tense mood of the track wouldn't be half as apparent. Pedal points can occupy any register – sustained bass notes, repeated high-string notes, vocal melodies or harmonies in the midrange – as long as that single pitch persists throughout. They're effective because what they're actually doing, with each chord change, is stamping a new identity on each new chord. Play an F note over the top of a C major chord, for instance, and the perceived result is the feeling of a Csus4 chord, and all the implied tension that goes with it. Let's get pedaling!

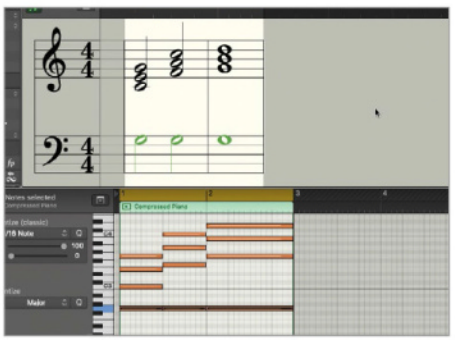
Step by step Working with pedal points



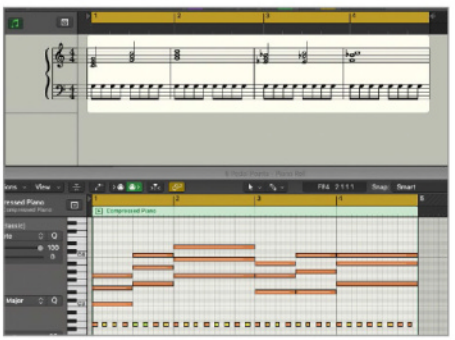
01 Here's a harmonized C major scale – seven notes, from C to B, played on the white notes of the piano keyboard. I've harmonized it by stacking up alternate notes from the scale in threes to make three-note chords, or triads, labeled with Roman numerals – upper case for major chords and lower case for minor and diminished chords.

02 The most obvious choice for a pedal point is the root note, or tonic, of the key we're in. Let's demonstrate by superimposing a high C note over each of the chords in our harmonized C major scale. Even though C doesn't appear in all the chords, it still sounds relatively harmonious, because it's the tonic note of the key.

03 Let's make a simple progression using some of these chords – this will make it a diatonic progression to the key of C major. Using the tonic I, IV and V chords, we can create the progression I-IV-V, or C > F > G in this key. Let's also include a left-hand part, playing the root note of each chord and following the progression as the chords change.

04 Now look what happens when we change the bass part, so that instead of following the chord changes, we just stick to one note – the tonic of C. Now the chords are effectively C > F/C > G/C – they've become what are known as slash chords, and the progression suddenly sounds a lot more exciting, as if something impressive is heading our way.



05 Another choice might be the dominant – the root note of a key's V chord. This creates a restless, unresolved feel as you're yearning to hear it drop to the tonic. To illustrate, let's take the same progression, but shift the sustained bass note to G – the dominant of C major. We're left with a feeling of wanting to move on to somewhere else...



06 ... So let's now take this further and add a few non-diatonic, 'borrowed' chords. How about tagging Eb major (Eb, G, Bb), Ab major (Ab, C, Eb) – both of which are borrowed from the parallel key of C minor – and Bbsus2 (F, Bb, C) onto the end, but continuing with our sustained G in the bass? And while we're at it, let's change the pedal point to eighth notes.

Recommended listening

ZEDD ft TROYE SIVAN - PAPERCUT
Clever one, this – Zedd has effectively remixed his own song by the end, the two sections linked by an insistent F synth note.

bit.ly/zedd_ppc

KATY PERRY – NEVER REALLY OVER
The chorus melody is a repeated Bb over an Ab > Eb > Db > Eb > Fm > Cm > Db > Eb progression. The key is Ab major, so the pedal point is actually a repeated 2nd**.**

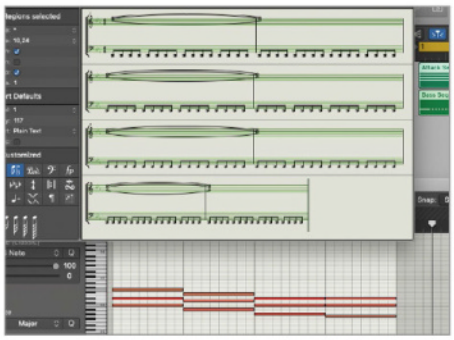
bit.ly/kp_nro

Pro tips

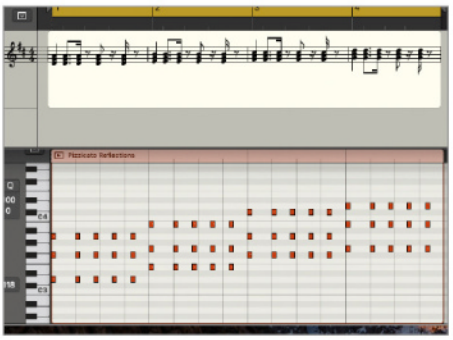
Bass moves
Next time you're putting together a chord progression and a bassline, instead of slavishly following the root note of the chord with the bass, try keeping the bass note the same through one or more chord changes, or change it early to announce the forthcoming chord before the change happens just to mix things up. This was done to good effect on Ariana Grande's 'Into You'.

Find the pulse
A classic technique in dance is to sustain a long, high-string note over a breakdown. You can enhance the effectiveness of this by strapping a compressor across the string and sidechaining it from a kick drum whose output has been removed from the mix, solely for the purpose of triggering sidechains. This can be done in Logic by setting a dedicated kick drum track to 'No Output'. This makes the string pulse invisibly wih the tempo of your track, maintaining the rhythm.

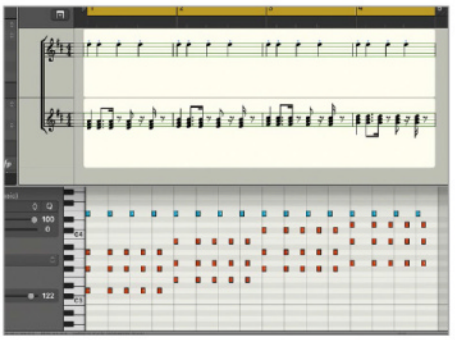
By Dave Clews
Over the course of his 25-year career, Dave has engineered, programed and played keyboards for numerous artists including George Michael and Tina Turner



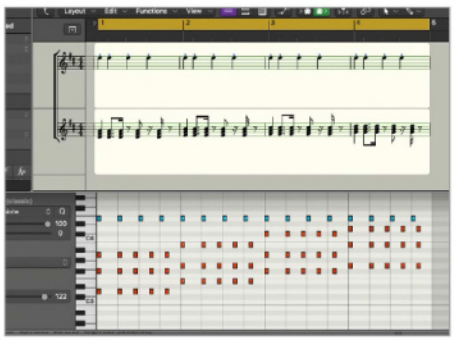
07 When working in minor keys, the 'pedal point emotion injection barometer' swings from excitement over to tension, especially when using a growly, sustained synth bass tone. Here's Curve 2 CM's Bass Syn Open A LE preset playing a repeated 16th-note C note under a moody C minor synth chord progression of Cm, Bb, Ab, Cm.



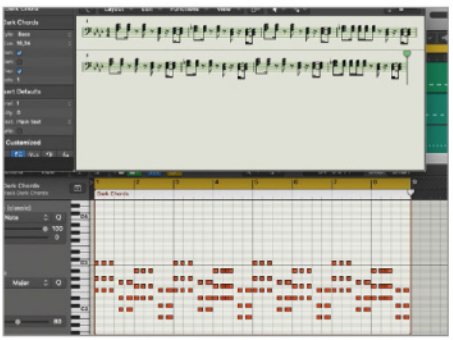
08 If the pedal point is a note that isn't the tonic or dominant, the discord this creates can enhance the effect even further. In fact, you could go crazy and use any note from the parent scale of the key we're in. Here's a D major progression I > ii > iii > IV, translating in this key to D (D, F# A), Em (E, G, B), F#m (F# A C#), G (G, B, D).



09 Let's add a repeated 2nd – the note E – as a pedal point, playing quarter-notes on a high-string sound. E is only found in one of our four chords, and it's neither the tonic nor dominant – it's the 2nd note in the parent scale of D major. Adding a high E to the first chord of D major (D F# A) effectively adds a ninth, audible as a D9 chord.



10 Adding the E to the second chord, Em, simply reinforces the chord's root note of E, but with the third F#m chord, we've created an F#m7 sound (F# A C# E). Lastly, E is the sixth in a G major chord, so this chord effectively becomes a G6 (G, B, D, E). Our apparent finished progression is now D9 > Em > F#m7 > G6.



11 An alternative pedal point technique bends the rules a bit by using an arpeggiator to generate a single-note sequence. To illustrate, we have an eight-bar section of a typical classic house track. The chord progression consists of all minor chords, exploiting the 'chord memory' effect popular in the genre – we've got Fm > Cm > Ebm > Bbm.

12 For the pedal point, I'm using an edited version of Dune CM's Amnesia 89 RL – an arpeggiated preset that I've tweaked so that all the arpeggio steps are playing different octaves of the root note of F. An automated filter sweep across the whole eight bars keeps things moving, with those insistent Fs conveying a sense of urgency throughout.

BLAST FROM THE PAST

Roland Jupiter-8

A lumbering behemoth, Roland's answer to the Prophet-5 became a classic in its own right, helping to define the sound of the '80s

Just as we put finger to key on this month's Blast From the Past, Roland announced the SE-02, a fully analog entry in their popular Boutique series and a collaboration with US analog mainstays Studio Electronics. Surprisingly, Roland and SE have chosen to overlook the larger company's legendary past catalog and have instead cobbled together something that is very, well, American, bearing some resemblance to Studio Electronics' own Minimoog-inspired heritage.

Despite the whelps of surprise currently echoing across synthesizer forums, this isn't the first time Roland have turned to US ingenuity for inspiration. Indeed, in its initial incarnation, one of the company's most coveted classics was designed to compete with instruments from across the Pacific, and its now-legendary sound had much in common with them.

That instrument is the Jupiter-8, a synth that's now seen as one of Roland's crowning achievements. Though relatively few units were sold at the time, the JP-8 has since come to exemplify the pinnacle of polyphonic perfection.

Roland's first volley in the war of polyphony was launched in 1978. Dubbed the Jupiter-4, it was relatively limited compared to the Sequential Circuits and Oberheims leading the charge. Though it cost far less than those instruments, it was quite expensive for a four-voice, single-oscillator instrument, and sales were less than stellar back then.

It would take Roland three years to offer musicians a truly viable alternative to the likes of the Prophet-5. With eight dual-oscillator voices and sounds that could be split and layered across the keyboard, the Jupiter-8 had the voice count of a fully loaded Oberheim OB-Xa and was clearly positioned to compete with the USA.

The JP-8's architecture was not unlike that of its American cousins, with dual oscillators offering triangle, pulse, square, and sawtooth waveforms (and noise for oscillator 2), plus an LFO, a resonant low-pass filter (that, like the one on the OB-Xa, could be switched between 12 and 24dB) and a pair of four-stage envelope generators. As Roland often did, they stuck a non-resonant high-pass filter in for good measure.

64 preset slots and eight combination slots made the Jupiter-8 an ideal tour companion (even if the poor tuning stability of the earliest models did not), and many a big name hauled the beast to stage and studio. Devout users included Greg Hawkes of the Cars, Tangerine Dream (natch!), Bryan Ferry, Thomas Dolby, Rush, Barry Andrews (Shriekback), and Ron Mael of Sparks. Michael Boddicker slathered JP-8 textures all over Michael Jackson's *Thriller*.

The sound of the Jupiter-8 was the epitome of 'analog mojo', and it's nigh on impossible to precisely reproduce. — hence the insane prices for second-hand units. Just as they did 30 years ago, those who want a bit of Roland magic for less money often make do with one of Roland's many follow-ups and spin-offs from the line. The rest of us look to samples and the many virtual polysynths inspired by the mighty Jovian Giant.

TECH SPECS

Year of manufacture 1981-1984
Original sale value $4,700
Current price $7,000+
Number made approx. 2000

Two plugin Jupiters

Roland Cloud **$30/MO**

Roland have attempted to bottle the essence of the Jupiter-8 a few times over the years, always tied to a hardware interface. Their most recent recreation comes in the form of a VST/AU plugin available through their new Roland Cloud subscription service and integrates with – but does not require – their System-8 Plug-out hardware.

www.rolandcloud.com

Arturia Jup-8V **$149**

As with previous Arturia clones, the pedigree of this Jupiter-8 emulation is now being thinly disguised. Don't let the truncated name and absence of Roland's logo fool you – this cross-platform plugin has all of the features that made the Jupiter-8 a pleasure to patch and play.

www.arturia.com

TOOLS OF THE TRADE

WITH A LEGACY FOR AUDIO EXCELLENCE THAT SPANS 75 YEARS, AKG products have captured the essence of the world's most iconic artists, influencers and podcasters. From our legendary C414 condenser microphone, born in 1971 and a staple in top studios everywhere, to modern-classic headphones like the award-wining K371 and the budget-friendly K240, we provide the tools to take your project or performance to the next level. Create your own legacy with AKG.

75 AKG YEARS

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Roland

Juno-X

$1,985
roland.com

By Dave Clews

Strengths

+ Oodles of original Juno mojo in a solid chassis
+ Stunning looks and familiar layout
+ Immense, authentic sound

Limitations

- Tiny screen can be confusing
- Speakers are underwhelming
- Steep learning curve for in-depth sound design

Roland's long-awaited Juno for the 21st century is finally here – but is it a worthy successor to the much loved original?

Roland's Juno-106 analog polysynth, launched in 1984, holds a cherished place. One of the first truly affordable MIDI-equipped polysynths, and an acknowledged cornerstone of '80s music, the 106 remains sought after.

With the current market price for a decent original Juno-106 around $2,000, what does the Juno-X offer for similar money? Well, nestling under that beguiling exterior lies not just a faithful digital recreation of an original 106, but also the next-generation Juno-X engine, a Juno-60 model, hundreds of sounds from the classic XV-5080 module and RD piano series, a vocoder section, plus the option of more downloadable Roland Model Expansions.

The Juno-X certainly has the wow factor. It feels heavy, the dimensions are comparable to the Juno-106, and it looks stunning. The classic red, white and blue livery immediately identifies it as a new arrival in the Juno family, with the "Programmable Polyphonic Synthesizer" legend. The chassis is solid aluminium and even the chiseled plastic end cheeks are faithful to the original, making the unit easy to lug around despite being built like a tank.

The front panel is bristling with sliders and multicolored illuminated buttons, in a layout reassuringly familiar to any previous Juno user.

Meanwhile, there's a proper IEC mains connector, balanced and unbalanced stereo audio outputs, a stereo audio input and traditional 5-pin MIDI IN/OUT ports. There are inputs for expression and sustain pedals, a mic input for the vocoder function, and a pair of USB ports for backing up settings to a USB stick, or connecting to a computer for use as a MIDI controller or a 14-input, 4-output audio interface. There's also two headphone ports and built-in speakers, through which you can stream music from a mobile device via Bluetooth.

Juno the score

The Juno-X's 61-key keyboard is a high-quality synth action drafted over from the Fantom-0 series and is wonderful to play, with a premium feel and just the right springiness to facilitate accuracy on fast runs. Channel aftertouch is present, although you really have to lean in hard to activate it. Anyone who's ever played an original Juno will enjoy the fact that Roland's combined pitch bend and modulation lever has not only survived, it's actually an improvement over the original, with more depth of travel in the modulation plane. More performance control is offered by two adjacent sliders and three switches that can place access to any parameter you like in easy reach of your left hand.

EXPANDABILITY

The Juno-X is an extremely capable and versatile instrument. If you simply want an updated version of the Juno-106, it can be just that, but there's much more power under the hood. For instance, as well as the numerous bundled synth engines, the Juno-X has the ability to be expanded with extra classic synth models from the Roland Cloud stable, such as the SH-101, JX-8P, or even Jupiter-8, among others. This makes the Juno-X a kind of 'several synths in a box' solution, so much more than just a modern take on the 106 alone. Its wide gamut of preset tones make it capable of producing sounds from any synth-based genre over the past four decades, with more than enough scope for hands-on editing thanks to the familiar controller layout. Roland Cloud subscribers can also purchase the WC-1 wireless dongle that can be used to purchase extra sound banks, expansions and other ZEN-Core-based Roland synth models. These can be transferred permanently to the Juno-X, with the only caveat being that you'll then need to reconnect to the server every 30 power cycles in order to reauthorize the licences.

Elsewhere, there's a brand new Supersaw oscillator type and three different filter types to choose from, (labelled R for Roland, M for Moog and S for Sequential we're guessing), plus there's an authentic stereo chorus with a new third mode that combines the effect of the first two. There's also a copious selection of modern multi-effects on tap, with dedicated hardware controls for reverb, drive and delay. Finally, the I-Arpeggiator imbues your playing with intelligent input based on the pitch or rhythm of your playing.

There are apparently over 4,000 preset tones in the Juno-X, including authentic digital recreations of the original Juno-60 and 106 factory patches, and sound banks from the XV5080 module, Roland's flagship synth module from the early 2000s, As someone who used a JV-1080 extensively in the past and still owns an XP-80, we can attest that these are very faithful recreations. You also get all the classic Roland TR-series drum machine sounds to boot.

Plug a dynamic mic into the port on the back (there's no phantom power support for condenser mics) and activate the vocoder model and you'll be rewarded with a real treat for the ears. The vocoder sounds incredibly clear, full and wide, although not as close to the tone of

the VP-330 as you might expect. There are only two sounds available – Vocoder 1 is the brighter, more digital-sounding of the two, while Vocoder 2 has a warmer, fuller sound but is no less distinct for it.

The original Junos found themselves heavily favored for bass and pad sounds, so it's no surprise to find a lot of the available sounds in the Juno-X devoted to these categories. Brassy pads, lush strings, rubbery plucks and plonky basses can be found in abundance, and flicking through the list readily brings a smile to the face of any Juno aficionado, while the Juno-X engine backs these up with a selection that includes edgier, more modern sounds.

The Juno-X doesn't score too highly on 'approachability'. Sounds that you play from the keyboard are called tones, and these are recalled and stored in a part, together with settings such as pan and EQ. You can then layer up to five parts together into a scene.

Scenes also store sequence data, effects and I-Arpeggio settings, with complex scenes sometimes reminiscent of the auto-accompaniment mode on an arranger keyboard.

On power-up, the Juno-X defaults into Scene mode, so individual sound selection is not

obvious. We had to embark on a substantial learning curve to figure out how its sounds and patches are structured.

The ZEN-Core synthesis technology under the hood means a generous 256 notes are on tap polyphony-wise, although polyphony does vary according to which engine you're using. There's a TR-style step-edit mode for sequencing I-Arpeggio patterns, especially useful for use with the built-in drum sounds. This transforms the two rows of buttons into step entry buttons, à la TR-808.

> "Flicking through the list readily brings a smile to the face of any Juno aficionado"

Juno how it is

The front panel controls are only relevant to the selected part, but you can select multiple parts at once and have the sliders control everything together. The layout as it stands is tailored towards the Juno-X engine, and labeled functions will change according to the selected synth model, but touching any slider will bring up its current function and value on the central screen. The ability to tweak parameters individually on the fly is a wonderful thing, and the sliders themselves are high-res, 1,024-step units.

Issues? Well, the slider controls are tricky to make out under dim stage lighting conditions. The built-in speakers lack low end and are really only suitable as a last resort if you have nothing else available to monitor through. The tiny screen often appears congested, considering the work it has to do to effectively grant access to the many hundreds of parameters contained within a synth of this depth.

However, there are two different software editing options available for the Juno-X. One is Roland's dedicated free Juno-X editor, which syncs with the synth in real time over USB, turning any available computer into a large colour display, while the other is Roland's Zenology PRO software synth plugin. At least the screen is in the optimum position on the Juno-X, rather than the bizarre upper-left corner location Roland used for the design of the Jupiter-X.

Despite appearances, then, there's nothing truly analog about the Juno-X. Some would argue that it's simply another fancy hardware container for the ZEN-Core engine. Yet the Juno-X is somehow much more than that, connecting with the user on another level that makes it more than the sum of its parts.

Roland claim that it fuses authentic vintage spirit with modern creative power, and we'd have to agree, but despite definitely needing to reach for the manual to get the most from it, to have this machine under your fingers, looking and sounding the way it does, is undeniably inspiring, particularly if synthwave or '80s-tinged electronic pop is your bag.■

THE ALTERNATIVES

ROLAND
Jupiter-Xm
$1,500
Roland's ZEN-Core-based modern derivation of their legendary Jupiter line squeezed into a compact, portable battery-powered format. Throw in another $900 if you want the full-size Jupiter X.
roland.com

ROLAND
Fantom-06
$1,385
Another ZEN-Core-based synth from Roland, but with more workstation capabilities and a color touchscreen.
roland.com

ROLAND
System-8
$1,385
Roland's flagship 'Plug-out' synth is built around its Analogue Circuit Behaviour tech (found in most Boutique synths), arguably surpassed by the newer ZEN-Core engine.
roland.com

Apple

MacBook Air M2

from $1,436
apple.com

By Andy Jones

Strengths

+ The lightest and thinnest Apple laptop
+ Powerful and silent
+ Lovely screen and speakers

Limitations

- Not a huge step up from the M1 Air for audio creation

Apple's latest laptop is the company's lightest and sleekest. It's also quiet, and could be the music production laptop we've all been waiting for

Apple's MacBook Air M2 2022 is a machine that delivers the battery life, speakers, power and, importantly, the silence you need for both music production and listening. You'll be paying for it, of course, but the Air M2 is also one of the most elegant and refined pieces of Apple technology ever created. Quiet, powerful and so sleek, it'll handle your music production needs with no fuss or bother. And obviously it looks amazing.

To Logic in no time

We admit it, hands down, unboxing an Apple product is one of the most joyous experiences in life. It is legendarily such an 'event', though, that the initial unveiling almost always outshines the end result where you end up using your brand new and much costlier Apple (iPhone/iPad/laptop/desktop) in the same way you used your old one – inefficiently, with all the baggage and the same old apps.

But that's perhaps not necessarily the case with the new MacBook Air M2 2022. Right from transferring your old data to it (a less than 50-minute task by Bluetooth) to making music and listening to the results, this feels – finally – like a new chapter of Apple use, and one that will drag your laptop music making to another level.

After a migration from our old M1 Mac Mini, we have Logic and Ableton Live up and running and playing old songs in no time at all. One or two downloads and a restart later and all of our plugins are there. And they are there fast. Sure, this isn't so much to do with the MacBook Air, per se – more an easy path created by Apple – but we feel all this business should be more hassle... just because it usually is.

Real-world use

At just 1.24kg, this is Apple's lightest machine and also their thinnest, just 11.3mm deep. You can hold it with two fingers and the all-new colors will have you showing off even more (we opted for the light gold Starlight, although the Midnight finish was also a bit of a temptation).

The sound emanating from its four speakers (which also support Spatial Audio) is so much better than a small form device like this should be capable of. You're not going to be using it for detailed mastering, granted, but it's almost good enough to mix with and very decent just for music listening. And for more detailed mixing and mastering,

> "The sound from the four speaker system is so much better than a device this size should be capable of"

just plug in some decent headphones and you have the perfect lightweight mobile studio.

A pro price

The M2 MacBook Air is undoubtedly expensive, so much so that you can only really compare it to Apple's latest 13-inch M2 MacBook Pro. Our test model Air M2 (24GB RAM and 1TB drive) costs $2,470 – exactly the same wodge of cash as the same-spec'ed M2 MacBook Pro.

And if you go down the MacBook Pro route you also get some performance advantages, although these are not perhaps as big as you might expect. Our sister site Tom's Guide tested the M2 MacBook Air directly with the equivalent M2 MacBook Pro and found quite similar results, with the Pro just nudging ahead on some tests. However, it's also a slightly thicker and weightier laptop with a tad smaller screen, not to mention an old design compared to the new, slicker Air.

This Air also has a better 1080p FaceTime HD camera and screen resolution. All things considered, then, we can say we'd rather opt for an M2 MacBook Air over a 13-inch M2 MacBook Pro.

So can we conclude that we'd choose the new M2 Air over any other laptop out there? Almost. There is just one other option to consider and that's the older M1 MacBook Air. It still packs a great punch and will give you almost as much power for quite a bit less cash. Ultimately, though, we think the M2 MacBook Air is beautiful, powerful and just about the best silent laptop you can buy for music production. After making music on laptops for more than 20 years, the Air M2 is finally the music production laptop we've been waiting for. Pricey, yes, but it's a powerful and very cool machine. ■

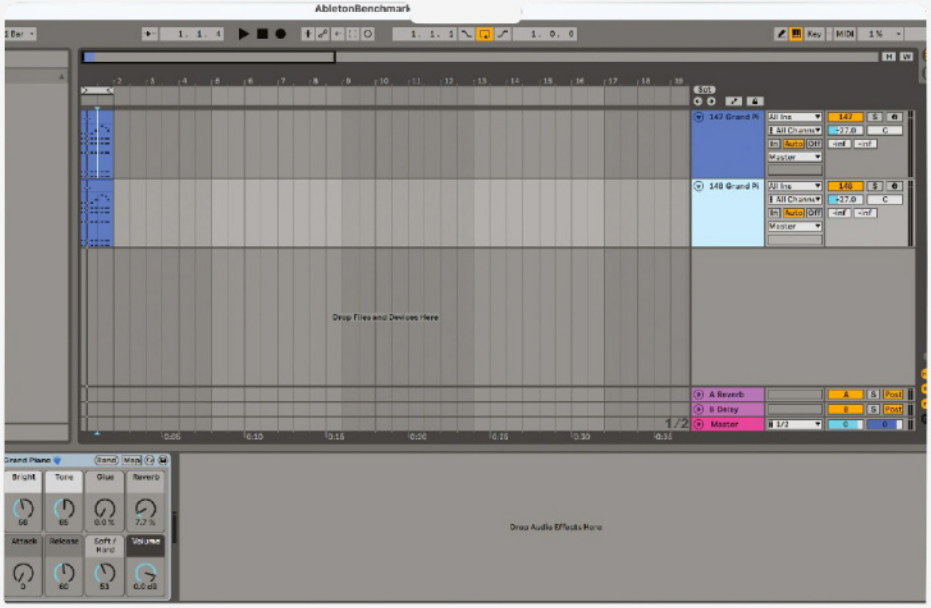
M2 v M1 v Intel!

So how does the new M2 MacBook Air compare to an older M1 Mac? First we did some tests using the same Logic project on both. The M1 booted up and loaded the song in 12.26 seconds while the M2 did it in a significantly quicker 10.75 seconds. When we then bounced the same track to audio, there was much less in it, with the M1 doing the job in 18.95 seconds and M2 creating the stereo file in 18.17 seconds.

We then turned to some more trusted tests and downloaded the Logic benchmark project from logicprohelp.com. With this, you load up the project and then add tracks on until 'System Overload' is achieved. Our M1 got to 92 tracks whereas the Air M2 got to 100, so it's an 8% uplift for the M2 over the M1.

With the Ableton Live benchmark project the M1 scored a track count of 136 while the M2 scored 148 so again, an 8% performance uplift for the M2.

While this isn't a huge uplift, we then did the test with a much older 2017 iMac (with an Intel core i5). It struggled to get to 34 tracks so the new M range has to be at least three times as powerful!



The Air M2 is Apple's lightest and thinnest laptop yet, weighing 1.24kg and measuring just 11.3mm deep

THE ALTERNATIVES

APPLE
MacBook Pro 14"
From $2,184
Probably still king of the Mac laptops with huge potential for RAM and storage upgrades. You'll pay for it though...

APPLE
MacBook Air M1
From $1,149
Not quite as much power but a lot less cash outlay.

Minimal Audio

Morph EQ

$49

www.minimal.audio

By Jon Musgrave

Strengths

+ Good choice of filter types
+ Flexible and simple morphing system
+ Global curve macro controls
+ Band-specific channel processing including mid and sides
+ Filter band auditioning

Limitations

- No individual bypass per filter

Bored with your regular EQs? This new Minimal Audio EQ with its morphing design could be just what you need to spice things up

Morph EQ (AU, VST, VST3, AAX) is a flexible EQ processor from Minimal Audio, and as the name undeniably suggests, it can be used for morphing effects, as well as more typical EQ duties. You get unlimited filter bands and seven filter types (high, low and band pass, high and low shelf peak and notch) with +/- 36dB gain where relevant. Main features also include series or parallel signal flow, optional gain dependent filter width, and an output limiter with both gentle saturation and punchy soft clip settings.

You have unlimited filter bands with independent channel assignment (stereo, left, right, mid or sides), but there's also a whole bunch of very handy global controls (Shift, Pinch, Spread and Scale). Shift and Scale both change all frequencies or all gains collectively, and are particularly handy from a time-saving perspective. Pinch, meanwhile, expands or contracts the overall frequency spread of the bands and is certainly interesting from a creative viewpoint.

Overall though, our favorite here would be Spread. For any stereo filters this lets you gradually offset the filter frequencies between left and right channels. Spread has a more marked outcome on more complex sounds, or where the filter curves are more dramatic (band pass for example). Nevertheless, we found that it was really suvvessful when it came to making creative changes to the stereo image.

Further features include frequency analyser display, context sensitive tooltips, global wet/dry mix, three gain scales, individual filter auditioning, input and output gain (-inf to +10dB) and undo/redo for filter changes. One thing that has not been included is individual band bypass, however we understand that this may be added at some time in the future.

Mighty Morph

Given its creative ambitions, it should come as no surprise that Morph EQ has plenty of presets showcasing its capabilities, and most of them include morphing pathways. Of the 100 or so presets, a few cover standard tasks (Tilt Shelf or Telephone Vocal). Thankfully, the vast majority of them are creative and grouped into four categories: Morphing, Abstract, Vowel and Filtering. And if you really don't know where to start, there's a preset randomize button, which picks a preset rather than randomizing the current patch parameters.

Morphing

The concept sounds simple enough, but how exactly do you set this up in Morph EQ? First of all, note that there is no onboard modulation. Instead a master Morph knob controls all active morph pathways. So if you want dynamic effects you need to automate this in your DAW.

Each EQ node (filter point) can be morphed and this is achieved using its halo. Click on a filter point and the halo will appear. Then drag from the halo to another point within the display to create a target. Starting from your new target point you can then create a chain of further nodes that mark out the morphing pathway.

Connections between target points are initially straight lines, but you can use your cursor to drag curved lines. Pathways can be deleted using the right click/CTRL-click menu.

Morphing pathways can be added to as many filters as you like, which ultimately makes for some incredibly complex outcomes.

> "We had great fun using Morph EQ to process basic white noise as well as long one-shot reverb tails"

Starting with the Morphing bank, presets such as Gyroscope and Low Swell deliver quite extreme changes once you move the Morph knob. Orbitz, as the name suggests, moves its four filters in a rotating circle from about 200Hz to 2kHz. Each preset delivers its own strange filtering effects as you move the Morph knob, and often you'll find a small section of the morphing path where the effect is just what you're after. This is ideal for creating special effects, and we had great fun using Morph EQ to process basic white noise as well as long one-shot reverb tails. What's more, a lot of these extreme effects can transform the stereo image with a tweak of the Spread parameter. Further fun can be had in the Abstract folder, which contains the most extreme effects, and in the Vowel folder, which has some great voice-influenced sounds.

EQ QED

Meanwhile, in the Filter category you'll find a mixture of traditional and innovative effects: Triple Highpass for example. It is, of course, completely possible to create complex animated EQ moves using DAW automation and a regular EQ, but Morph EQ takes that concept and makes it easier and more fun to achieve, and we think you'll love this wonderfully creative EQ. ■

Click on the headphones icon to activate the band solo audition feature

THE ALTERNATIVES

U-HE
Filterscape
$127
This excellent filter design includes eight morphable snapshots and onboard modulation

OBVIOUS
Filter
$59
Use the filter sequence and morph features to create excellent rhythmic filter sequences

vocaster two
48V
Host
Guest
Guest
Host
vocaster
DM2

Focusrite

Vocaster Two Studio

$540
focusrite.com

By Simon Fellows

Strengths

+ Well-designed, complete solution for amateur podcasters
+ Phone input, camera output and loopback are all hugely useful features
+ Plenty of clean gain for sensitive dynamic microphones

Limitations

- Combi inputs would have been a better choice than just XLR
- Pretty much dumbed down software

Everything you need to start podcasting, but is it everything you want? Let's find out

Spotify added 1.2 million podcasts in 2021, bringing the total number on its platform to more than four million. That's just one platform. Add in the other big hosts, and that's a lot of podcasts, and a good deal of new entrants, most of whom won't have any production skills. If, like them, you're more focused on content than technical production you're going to want a simple recording system that just works. Press a button, two at the most, job done.

Which is where Focusrite's Vocaster Two Studio comes in. As the name suggests, it's a neatly packaged podcasting kit aimed squarely at amateurs looking for a complete solution. Its sister kit, the slightly cheaper Vocaster One, is a single channel option for solo creators but, if you can afford it, the twin channel Vocaster Two will future-proof your investment. At some point you'll want to record a guest, surely?

It takes two

The kit comprises a two channel Vocaster Two interface, a DM14v dynamic mic, a pair of HP60v headphones, an XLR cable, a USB-C to USB-A cable and a portfolio of podcasting software. The interface, microphone and headphones all appear very pro in black, a look unified by a matching set of subtle red metallic stripes.

The obvious omissions are an additional mic, a second pair of headphones and cabling for a guest. However, the Vocaster Two Studio isn't cheap so I'm guessing these items would have

THE BUNDLED HEADPHONES

After enjoying the quality of the mic, the HP60v closed back headphones are a little underwhelming. The build quality is alright, they're comfortable enough and they do the job sound-wise. I just fail to understand why Focusrite has bundled a pair of on-ear cans when clearly an over-ear pair would do a much better job of isolating potential sound bleed. Perhaps it's because these offer a more universal fit, but they'd be the first thing I'd swap out for something more suitable.

THE BUNDLED MIC

The included DM14v is reminiscent of a streamlined Shure SM7B, arguably the best podcasting mic, bar none. Just like the Shure, it's a low sensitivity dynamic mic that's excellent at rejecting unwanted noise but does demand a lot of clean gain. As mentioned, the Vocaster has more than enough on tap. It's a basic XLR mic with no on-board EQ controls, but that's no bad thing here: there's no setup required. Just like the SM7B, it does feature a built-in pop-filter, windshield, internal shock mount and sturdy mount bracket that'll work with any stand or boom. It's a weighty beast too, with a tough all-metal construction. I was really impressed with the sound, especially the mid-range where the voice sits. Unsurprisingly for a dynamic, it's not detailed but has a flat response with little coloration and enough warmth.

pushed the price beyond the level most newbies are willing to invest.

The Vocaster Two's attractive pebble-shaped form is laced with seductive, tactile plastics and smooth, integrated buttons, something of a far cry from the brutal, rectangular metal bricks that we're used to seeing. Round the front are two 1/4" headphone outputs, one marked Host, the other Guest. Spinning it around, the back panel features two XLR inputs, two 1/4" TRS-balanced speaker outputs, a 3.5mm TRRS phone input, a 3.5mm TRS camera audio out, a 48V phantom power button, a Bluetooth button and USB-C port (USB-3).

The smooth, matt black top panel, which unfortunately picks up fingerprints too easily, has a giant central rotary input level control, flanked by two volume controls for the headphone and speaker outs. Sitting nice and tidily in a strip below these are found a mute, enhance and auto-levels buttons for each channel.

Basic operation couldn't be easier. Plug in your mic, hook up the interface to a laptop (or USB-C equipped iPad), launch your DAW, don your headphones, set the gain level and speak. Just like most interfaces in that respect.

However, the Vocaster Two has half a dozen standout features that make it particularly attractive to amateur podcasters. Auto-gain is a feature commonly found on interfaces, but nevertheless it's good to see it here because it removes much of the faffing and anxiety from getting a decent recording. Press and hold one of the two input gain buttons for ten seconds while speaking into the mic, and it'll then automatically set the level for that particular channel.

Talking of gain, the Vocaster can provide up to 70dB, which is enough to drive the most sensitive dynamic mics out there without resorting to a Cloudlifter. Podcasters favour dynamic mics for noise rejection potential, so this is a real boon.

Yakety yak

The phone input and output, either via cable or Bluetooth, is heaven sent. I can't think of a podcaster who doesn't occasionally interview guests over the phone, so this feature will be well used. Similarly, the camera out is a blessing for the many podcasters who also stream or post their podcasts to YouTube. It provides instant syncing and could prove a real time-saver.

Finally, the interface ships with Vocaster Hub software, which enables you to control all the aforementioned features in an intuitive, graphic-heavy, tech-lite way. It's also the place to fine-tune your levels for the best possible mix. Yes, the software may be easy to understand but it's a bit too dumbed down for my liking. For example, the gain level indicators will display green, amber and red for clipping, but there's no scale. Call me old-fashioned, but I like to know where -12dBFS is...

On the plus side, the software does reveal an additional feature, loopback. This enables you to record the output of your computer, which is useful for including everything from backing tracks, intros and outros to pre-recorded chats.

As well as the Hub utility/mixer the Vocaster Two Studio ships with Hindenburg Lite podcaster and radio recording software, plus trial subscriptions to Squadcast and Acast. Essentially, Squadcast is a cloud recording suite and Acast is a podcast hosting platform. Whether you decide to continue with these subs or not, Hindenburg is a solid piece of recording software, or you can, of course, use your own DAW. ■

THE ALTERNATIVES

UNIVERSAL AUDIO
Volt 276
$306
UA's new generation of audio interfaces comes in two flavors: the top-mounted-control of the '76 range works for podcasters and producers alike and you'll be glad for that extra compressor.
uaudio.com

T.AKUSTIK
Podcast Bundle
$234
Perhaps an ideal companion to the Volt 276 is this bundle from Thomann's t.akustik brand which adds all the extras you need to start creating podcasts.
takustik.com

TASCAM
Mixcast 4
$669
More than just an audio interface aimed at podcasting, the Mixcast 4 is designed as a multichannel mixer for all your podcasting needs. For more hands-on control, you'll need to stump up extra.
tascam.eu

BandH.com
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Audio-Technica AT2040 Hypercardioid Dynamic Podcast Microphone
AUAT2040 | $99.00

Focal Alpha 65 Evo 6.5" Mixing Monitor
FOALPHA65E | $449.00

Keith McMillen Instruments K-Mix Programmable Digital Mixer, Audio Interface & MIDI Control Surface
KEK737B | $599.00

Heritage Audio Baby RAM Passive Monitoring System
HEBABYRAM | $199.00

Avantone Pro Gauss 7 2-Way Powered Reference Monitors (Pair)
AVGAUSS7 | $799.00

Solid State Logic UF8 Advanced DAW Controller
SO726490X2 | $1,299.00

IK Multimedia iLoud MTM High Resolution Compact Studio Monitor (Single, Black)
IKIPILOUDMTM | $349.99

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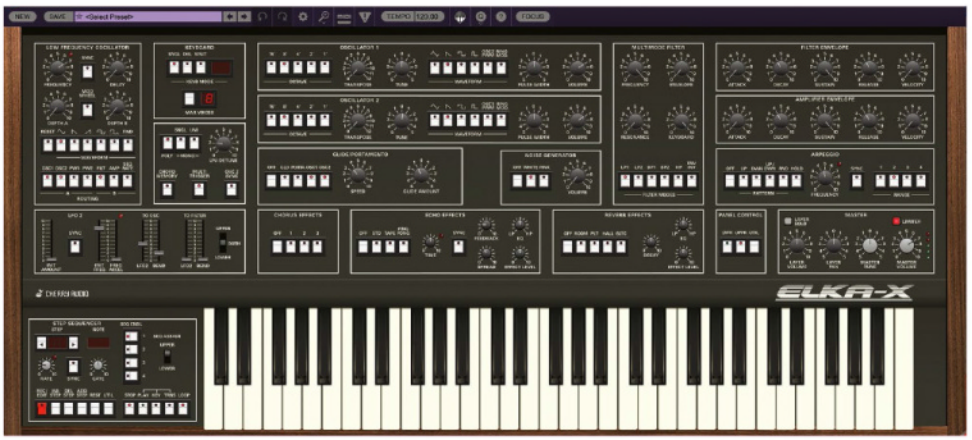
Cherry Audio

Elka-X

$59

cherryaudio.com



By Andy Jones

Strengths

+ Beautiful reimagining of a classic
+ The GUI is a total pleasure to use
+ Complete MIDI control through real time or DAW
+ Nice enhancements over the original
+ It's a total bargain at this price!

Limitations

- We can't find any!

The latest vintage synth re-model to enter Cherry Audio's lineup revisits an oft-forgotten European classic from the early '80s, the legendary Elka

There have been some fine examples of synths over the last 50 years that have been created by smaller companies. Brands such as the Oxford Synthesiser Company and EMS produced models away from the American and Japanese spotlight. Staying with European underdogs, Cherry Audio now visit the heart of Italy, where Elka became respected producers of keyboards, organs, and the odd synth.

The 'X' factor

The Elka Synthex was Europe's answer to the large polys of the early 80s; while it was intrinsically analog, it garnered a number of features, such as state variable filtering and cross modulation, which meant that it gave the early digital machines a good run for their money. It's a difficult and expensive machine to find as a second-hand prize, which makes the appeal of a good plugin version all the more enticing.

Cherry Audio's impressive roster of analog remodels in plugin form, boasts both synthesizers and effects. The all-new Elka-X embraces the very spirit of the Synthex, while making a few enhancements and improvements along the way. The first immediate enhancement comes with polyphony; the Elka-X doubles the original's voice count to a full 16 notes, while also offering split/multitimbral operation, like the original.

The architecture provides two oscillators per voice, originally DCOs, which initiated 80s tuning stability as a consequence. Reliability of tuning is included as standard here, but the feel and sound of the oscillators is exemplary. Even with a single DCO in play, there is a beautiful depth to the lower-mids which feels entirely natural. The filter continues this tradition, with a further enhancement; all the original options are included, from low to band and high-pass filtering, but with the addition of a grittier 12dB low-pass filter, to accompany the original 24dB.

Getting started

The Elka-X is packed with some really great presets. These are immediately accessible from the browser, and take full advantage of the Elka's complexity, right down to the inclusion of a programmable sequencer and separate arpeggiator.

However, one of our favourite features is the New button, found nestling right next to the browser area, which initiates the

> "Even with a single DCO in play, there is a beautiful depth to the lower-mids which feels entirely natural"

synthesizer to a default state. This is important, because it makes the building of new patches very easy and instantaneous, and frankly with a beautiful interface laid before you, as we have here, why wouldn't you want to create your own sounds?

Cherry Audio have catered for all the usual plugin formats, but there is a particular enjoyment to be had by just opening the standalone version, and losing yourself in synth nostalgia. While the Elka-X will immediately sync-lock to your DAW, there is inclusion of an onboard tempo function, which in turn synchronizes elements such as the sequencer, LFO and delay effects. You will also find onboard reverb and chorus, the latter being particularly classic and effective, and a calling card from the original.

Final output

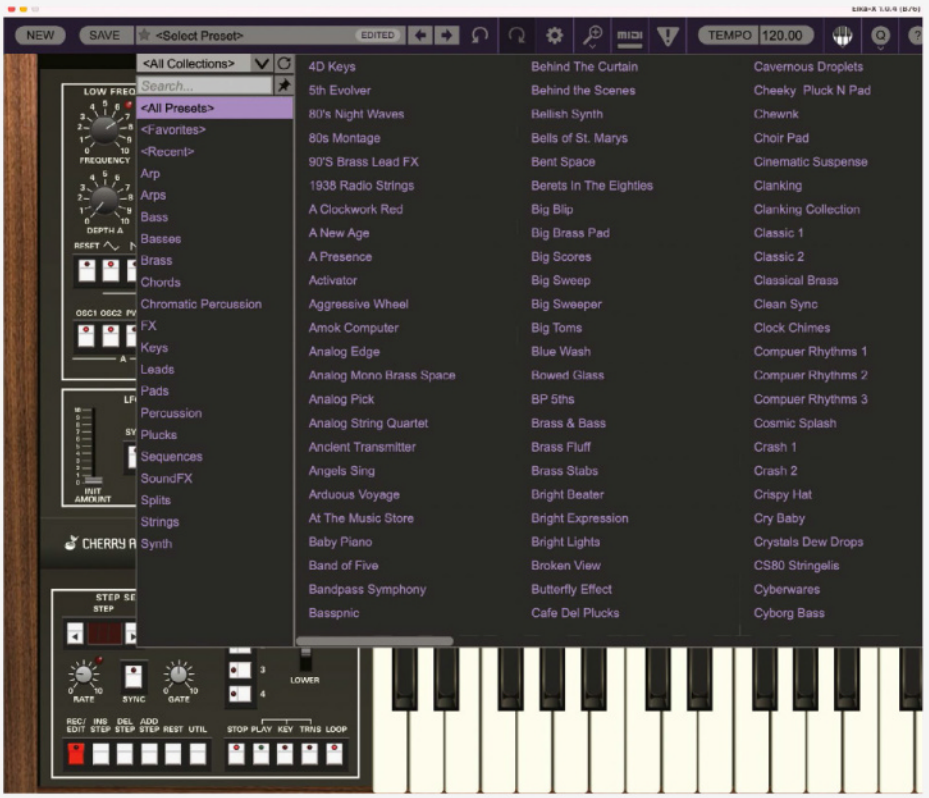
Cherry Audio are attracting a serious reputation for creating highly usable and great sounding re-imagined versions of some of the most desired synths from our history. The Elka-X slides gracefully into that arena, with a depth of sound that inspires creativity from the first note. Don't be fooled by its highly attractive price point; it's a bargain for sure, but with no sense of sonic compromise.

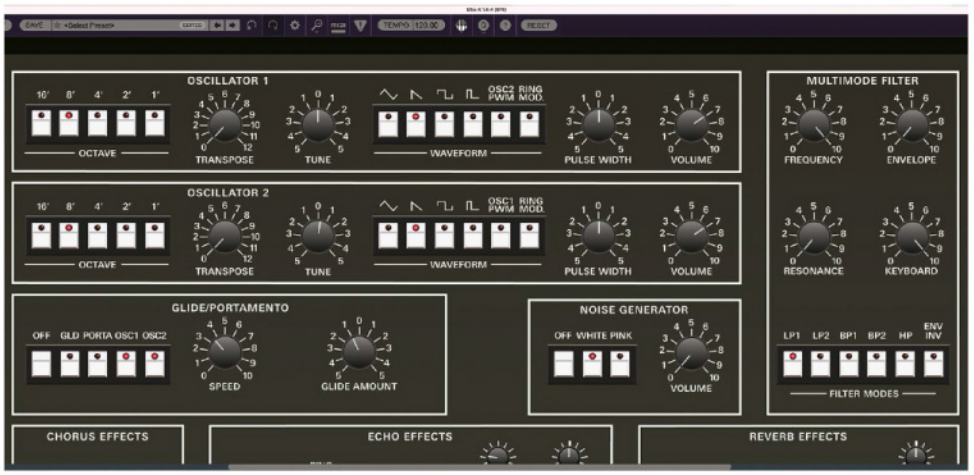
If this release doesn't shake up the market, you're not playing your saw waves low and loud enough! ■

MIDI control

One of the elements which we particularly enjoyed with this plugin is its standalone operation, allowing you to get creative away from the DAW. With the Elka-X grabbing the majority of your computer's screen space, it's the closest thing you will get to a hardware experience in plugin form, at least until we have the complexity of touchscreen that we all yearn for.

Until that day, we can make do with the facility provided by Cherry Audio, that allows the easy assignment of any of the GUI controls to MIDI hardware faders. This also extends to MIDI CC control from within the DAW, but it's nice to have the easy option to assign the filter cutoff to a hardware fader or pot. Of course, we don't all have large computer screens at our fingertips, so the Focus Mode will zoom right in on the area that you are working with, providing plenty of detail and fine control.





The beautifully crafted layout of the Synthex, zoomed in using Focus mode for finer detail

THE ALTERNATIVES

UVI
Synthex 1.5
$78
Inspired by the Synthex and the FM-based Elka EK-44, UVI's homage appears here in a sampled form

XILS-LAB
Syn'X 2
$169
Full multitimbral version of the synth with eight layers of lovely Elka-ness to play around with

IK Multimedia

MODO Drum 1.5

$288

www.ikmultimedia.com

By Andy Price

Strengths

+ Unprecedented kit and beat editing via modal synthesis
+ Great playability/humanization options
+ Simple UI and easy to use
+ Deep room shaping, kit-mixing scope

Limitations

- A real disk-space hogger!

With some of the most expansive tone-shaping control we've ever handled, IK have augmented an already unparalleled drum workstation

Beats. Everyone needs them, everyone tackles them in their own way and it's widely agreed that when it comes to mixing, they're more often than not the most awkward thing to get right. IK Multimedia's MODO Drum aims at serving all needs via its holistic, genre-neutral approach to kit control. Though the emphasis here is clearly on giving you the sound of a live, in-the-room drum kit, the sheer level of tonal shaping, FX, groove-shaping and real-world nuance that you can control here, results in perhaps the only piece of beat-building software you'll ever need.

From the outset, IK offer a smooth, straightforward experience, a quality for which the company are well known. MODO Drum brings the same multi-page, visually entrancing approach as MODO Bass. V1.5 brings three more kits to the original's ten, all realised with IK's detailed physical modeling drum synthesis.

Kit bag

The kits on offer here range from the six-piece Rock Custom, aimed at the vest-sporting, long-haired, seasoned kit vet, to the genre-neutral new kit for v1.5, Silver, which could be used across pop, country and rock. Then there's the cutting, razor sharp tones of the Djentleman, a kit directed at a particular niche of metal, but whose biting quality works well in a variety of other contexts. 1.5 also brings in other new kits not previously available: Brit Custom and a tasty Metal kit. Actual kit selection is swift, with an image of each kit selectable via the Model page, which shows a depiction of the kit in all its splendour. MODO Drum's array of seven core pages are neatly displayed at the top of the UI.

While the kits have been assembled to fulfil specific objectives, within the Customize page you can mix and match drum elements from other kits, and build your own Frankenstein kit to better suit your needs. That's not all though: each constituent part can have their virtual skins swapped, be retuned, have the buzz dialed up or down, and re-define the shell profile. This is all down to the mind-boggling synthesis technology that certainly trumps the conventional approach to capturing samples.

This flexibility is further evidenced by the breathtaking effect of changing your virtual drum room via IK's deft use of convolution, or perhaps effect-soaking certain kit elements in the Mixer.

Simulation theory

Some of MODO Drum's most extraordinary abilities, such as the option of having infinite round robins, is thanks to the real-time modal synthesis technology used to create these virtual kits. Unlike traditional sample capture, this innovative tech mathematically models the physical parameters (dimensions, skin, materials, durability etc) and all the in-between variables including play style, stick-type etc and room vibration to swiftly produce what is, in effect, a very real model of the kit within your CPU. You can affect it in real-time, without any processing lag or having to settle for approximations of what certain changes 'might' sound like. In effect, it's like tiny virtual engineers are shifting mics, swapping drum elements and more instantly within your hard drive. It's a technology that IK have been working on for 11 years, and the first version of MODO Drum, released back in 2019, marked IK's first commercial deployment of it.

Playstation

While the kit's sound is highly malleable, MODO Drum is also loaded up with a range of playability options. Going well beyond what we've seen elsewhere in the 'virtual player' domain, MODO Drum's Play Style option allows for the precise choosing of sticks, exactly where on each drum element the sticks are hit, and just how accurate this digital drummer's impacts are. Quite frankly the sculptural scope is astounding. The Grooves section is the easiest way to hear the results of all of this in full flight, with a pyramid of pre-determined drum patterns. As with MODO Bass, these samples are super easy to slot into your project.

Using MODO Drum to build up a series of demos results in a rewarding experience, and while we find it very easy to get entirely comfortable with the surface level offerings of the library, there's a veritable ocean of possibility beneath the surface. While size might be an issue, thankfully you're able to download only those kits you need. While other drum libraries trade off on bringing you accurate, playable recordings of premiere kits in world-class locations, IK instead hand the keys to you, with the player, room, kit and character entirely at your control. ■

Every drum component can be switched and re-tuned

THE ALTERNATIVES

TOONTRACK
Superior Drummer 3
$365
This powerhouse was previously our top pick for versatile kit-shaping and control

XLN AUDIO
Addictive Drums 2
$161
While a different approach to MODO, it's still a solid kit handler

UVI

Mission 6

$80

www.UVI.net

By Dave Gale

Strengths

+ Captures the Andromeda spirit
+ Plentiful supply of presets and samples
+ GUI styling is classy and reminiscent
+ Improved arpeggiator, over the original

Limitations

- Not entirely identical sonically
- Lacks several characterful elements, available with the original

Inspired by a lesser known, but respected analog classic, UVI's latest Mission visits the limits of Andromeda

As analog classics go, the Alesis Andromeda A6 is not the model that most cite as a favorite. Part of the reason for this could be that many original Andromedas have suffered component failure, placing them out of operative reach. But find one that works and you'll hear a huge-sounding analog from the year 2000, when analog wasn't really the flavor of the millennium!

T-minus and counting

Thankfully, while the pricetag of an original second-hand Andromeda remains high, UVI can help you with another one of their superb re-enactments, providing the very spirit of the original analog machine, through a vast number of samples.

Mission 6 resides within the usual UVI ecosphere, being operational through their freely available Workstation plugin, or for extended use via their Falcon synthesizer suite. For our money, the Workstation plugin presents the information in an incredibly concise manner, while being highly stable as a plugin. You certainly will not be missing out on the party by not using it through the Falcon software, although you can take the editing capabilities far further, if that's an interest for you. You'll need a spare 12GB of disk space, which not only confirms the sampled nature of this library, but also underscores the 400 beautifully curated presets, to get you going.

This is not to say that you cannot perform deep editing, because the Mission 6 is laden with the usual UVI suspects. The main edit page draws graphical inspiration from the original machine, and, in many respects, provides a more simplistic and usable interface. One criticism often leveled at the original machine is that it was incredibly complicated to use, largely thanks to its comprehensive modulation capabilities, and an LCD display which didn't always provide the strongest and clearest clues, while in edit mode.

Ignition sequence start

This library conforms entirely to the common UVI format, allowing the loading of up to two sampled sounds/waveforms to form a patch. The raw samples include absolute basics, like saws, squares and triangles, along with pulse-width modulated sources and distorted tones. There are also plenty of more colorful and programed samples, perfect for creating more elaborate and lush textures. These are all beautifully categorized, making light work of finding

Getting patched

We are very big fans of UVI here; they have a proven track record for creating interesting products, which are production ready, and good-to-go. Part of the strength in this remit is their curation of samples and presets, in a hierarchy which makes light work of the substantial content. If you like the idea of becoming inspired by presets alone, you will find plenty of basses, brass, leads, pads and polysynth sounds, all indigenous to the Andromeda. If you prefer to get busy with editing, there is equally plenty of initial content, providing the perfect launchpad for sonic creation. Collated within the Basics category, you can begin with a mono sound source, right through to an initialized Unison of waves. That was always an Andromeda calling card, as the 32 analog oscillators could be stacked to create the most almighty and super of saws! What's not to love about that?

> "Represents the very spirit of the original most admirably in an easy-to-use and immediate incarnation"

something suitable. The 'Main' working page is also where you will find the filter settings, providing low, band and high pass iterations. These are unsurprisingly not quite the same deal as the Andromeda, in part due to the complexity of the filter section included with the original, but are perfectly functional, with four-stage envelopes to accompany both the amplifier and filter sections. Four other pages give considerable modulation, editing and effects capabilities, along with a comprehensive arpeggiated section.

Mission Pros and Cons

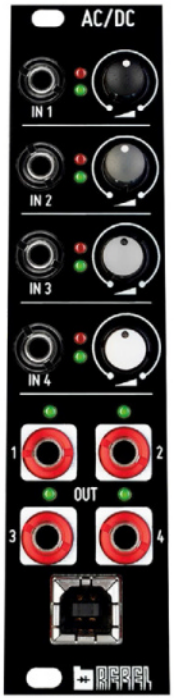
UVI's exemplary presentation of the Andromeda in this form provides a great sounding and production-ready alternative to the original.

Many would convincingly argue that the strength of the Andromeda lay with its preset patching, with the ability to tweak at will. While the Mission 6 lacks certain elements (albeit sometimes incredibly complicated ones), what it does offer is a very healthy facsimile, which is inspirational, ready to go, and sounds very fine indeed. Mission 6 represents the very spirit of the original most admirably, in an easy-to-use and immediate incarnation, and lest we forget, it is quite literally a fraction of the price! ■

THE ALTERNATIVES

LFO AUDIO
Andromeda A6
$45
Packed with an original set of Andromeda patches, this is a high-quality sampled emulation

SPECTRASONICS
Omnisphere 2
$363
Includes numerous Hardware modes and a template for Andromeda simulation



Befaco

AC/DC

$240
befaco.org

By Rob Redman

Strengths

+ Takes little rig room
+ Intuitive workflow
+ Opportunities for development

Limitations

- Limited patching IO

Did this DC-coupled audio interface from Spain, hailed for its relative affordability, shake us all night long?

For every patch there is the final question; how to send the signal for output. Use a mixer then head for a desk, headphones, or speakers? Multitrack into your DAW of choice? Or both? While there are some single module options for this, they tend toward the larger, more costly devices, which may not suit the rig or budget of all users. Befaco's AC/DC may just be the solution for many modularists.

The module itself is pretty diminutive for the features on offer, at just 6hp. The build quality is good, although maybe not quite up there with the most robust options on the market. That said, it is available as a prebuilt or a DIY module, which will give some confidence to those who are happy constructing their own gear.

The physical unit is small, neat and clear. There are four outputs and four inputs, each with a level control and clipping LEDs, making it easy to ensure levels are as required. Plus a USB type B connector for data traffic in both directions. That's it for the front panel – other than simple, clear labeling, something all modules should have, shirking the hieroglyphs some developers opt for. It's plain – and some may say dull – but it sure works.

The AC/DC has a few strings to its bow that really make it more than the sum of its parts. As the name suggests, it is DC-coupled, meaning it can act as a bridge to your computer, not just a way of getting audio in and out of your rig and Mac or pc. If you use Ableton Live, you can send, via USB, modulation and sequence data in either direction, courtesy of Ableton's CVtools. This may not be of use to all but it is worthy of consideration when appraising any IO needs for your rig. It opens up a world of possibilities for all manner of musical situations, from simple multitrack recording to using software plugins as audio processors. If a rig has limited room but a patch could do with some reverb and compression for example, the audio can be sent out to software like an effects loop.

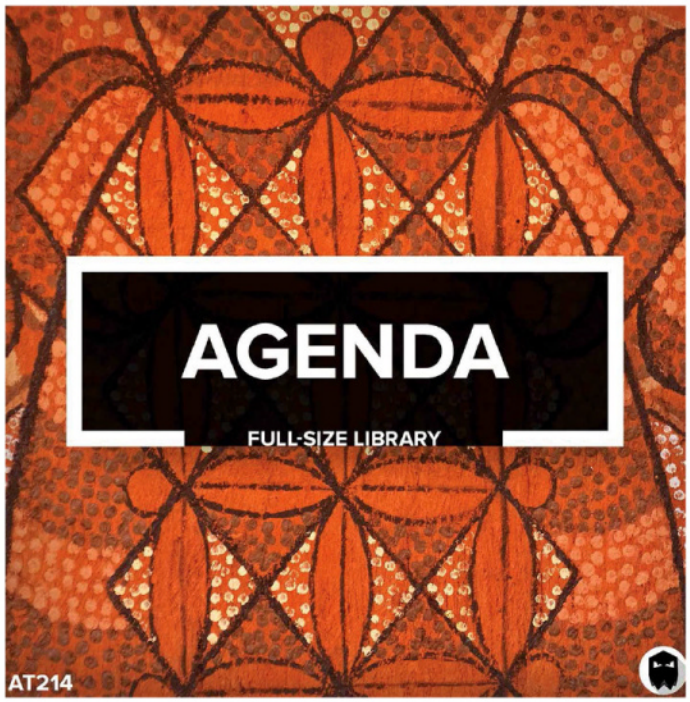
Lastly, the AC/DC runs on the OWL platform, opening up the module to development by any user with the desire to do a bit of programming. The AC/DC OWL user community is in its infancy but expect to see keen enthusiasts sharing as time passes.

The DC coupling can't be overestimated here, as it really does open up a lot of doors. Not just for audio processing but for the CV control mentioned earlier. For users with a favorite sequencer, be it DAW-based or as part of their Eurorack rig, it's a godsend. No matter the sound source, the ability is there to send patterns in either direction, combining the best of both worlds and massively increasing the power on tap. USB-C would be nice, simply as it is becoming the standard, but the Type C connector is more than fit for purpose and can feel more secure, so not all bad. ■

Audiotent

Agenda

$56



Sun-kissed and fun-packed collection of riffs and rhythms inspired by the afro-house sound. Plenty of proper old-school synths and hardware machines get dusted off to provide some great grooves in this library and sit pretty with lively recordings of real instruments, for some tasty and energetic percussion.

The drums come in jackin' full form, just tops, and strong individual hits, and they work a charm with the rhythmic textures, synths and basses, and other merry melodies and musical loops.

The magic in this set comes from the sweet spot found in the mix between the expertly processed analog and the organic samples, making sure the club-rattling punch is there for your productions, alongside the playful dynamics you get from capturing live musicians.

Roy Spencer **audiotent.com**

Monstercat
Eminence
$14.50

Marvellous melodic house and techno selections from the studio of the celebrated Canadian production partners, DLN and Kazmo. This dancefloor-focused collection has loops running from 120–124bpm, taking in big bossy basslines, cheeky percussion, heavyweight drum hits and loops, and a sizeable stock of main top and musical grooves. The 239MB of built-from-scratch beats and pieces all have the duo's trademark flair for mood and movement, and offer fresh twists and turns on that digital chart troubling Eminence-brand of proggy dance music majesty. So, if you want a big room cinematic sound, at a pocket-conscious price point, try it out.

Roy Spencer **loopmasters.com**

Blind Audio
Freeform House
$8.80

Cool, catchy house samples, pulling inspiration from tough and techy dance music. IDM and electro noises make up mix 'n' match melodies. Coaxed carefully from a fat stack of iconic drum machines, legendary synths, vocoders, and other lusted over hunks of history-making hardware. Everything gets its circuitry warmed nicely, resulting in 329MB of covetable content locked to the classic club key of Am, and the party-provoking bpm of 124. All the perfectly polished and processed sounds are a doddle to drop next to one another. And every last bumping bassline and ear-worm hook has plenty of pin-point drum hits and loops.

Roy Spencer **loopmasters.com**

Splice
Dëra Sound Kit
From $7.99/month

Slick beats and wicked rhythms form the first instalment of this new sample series, tailor-made to empower the next young wave of African artists. To kick off the collection, a spotlight is on the emerging talents of hit-crafting Afropop producer, Dëra. Amongst his handmade loops, one-shot sounds and versatile presets are plenty of deep bass chords, weighty pads, finger-snapping grooves, and tweaked and twisted synth and FX work. And his gift here for Afrobeat drums, hats, shakers and percussion, only adds an insatiable swing on top, further lending to the infectiousness of each radio-ready hook. Summer-blessed stuff, and a strong opening for the new collection.

Roy Spencer **splice.com**

Sample Magic
Pounding Techno
From $7.99/month

Sweat-box soundtracking samples get cranked to 11 in this subtlety-free set of gloriously punishing techno. Tough-as-granite drums rain down like tornado debris, as swirling FX and pads circle wildly around window-shattering bass and drums. Not suitable for producers looking for loops to lift their next lazy pool party or BBQ groover: this is the strong stuff. And if you're the sort who dances on tables rather than hiding underneath them, you're gonna wanna crowbar the lid off this little lot and plunge your hands in as deep as they go. The 293 loops and 181 one-shots are made to menace with 909s and the like kicked and punched until they tell you exactly what you want to know.

Roy Spencer **splice.com**

MODERN DIGITAL PLUGINS

Get to know the past year's most forward-thinking plugins with this tour of the state-of-the-art

Many of us have been witnesses to a golden age of music tech, one in which digital music tools have come of age, developed their personalities, honed their craft, and become stone-cold masters of audio jiu jitsu. In the last decade or two, digital tools have gone from barely passable, to completely viable, to indispensible and widely adopted.

Last issue we explored some of the most innovative software based on 'old' sounds: analog circuit components, amp simulators and reverbs. While the technology of these are sopshisticated enough to make a time-traveling 1970s mixing engineer fall to their knees, the tools would still have been familiar and contextual, for the most part. This issue we're looking in the other direction: at the tools that shine a light on the next steps of audio history. What will be the next Melodyne or Auto-Tune? Which bold new tools will become indispensable? What software releases will change the face of music and inspire whole genres to come into being? Well, we can't quite tell you... but we can give you an idea for the cutting edge of today's digital technology by shining a light on a few of the newest and most interesting virtual devices to appear in recent months. Ready to see the world of tomorrow? Buckle in and prepare for countdown...

Waves Harmony

Since Waves hit the scene all the way back in 1992, they've earned a place at the head of the audio software table with an expansive, everchanging line of signal processing tools.

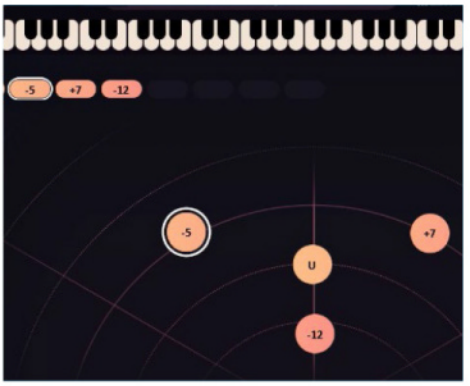
Regardless of which bag of music production tricks you wish to rummage through, the developer has no shortage of options. Mixing and mastering, noise reduction and restoration, sound design, plugins for live sound – all corners of the audio world are covered, and in pretty exhaustive fashion.

One of the areas in which Waves specializes is vocal plugins, having released a plethora of premium tools that handle tuning, de-essing, double-tracking, creative vocal effects and more. Their latest arrival at the party is Harmony, a "real-time vocal playground" that creates instantaneous harmonies and chords from solo vocals. Complex vocal harmonies can be gorgeous to listen to, and are usually the preserve of virtuoso musician types like Jacob Collier, Kanye West and Bon Iver, but Harmony aims to make things a little easier for the blue collar producers among us.

The plugin can generate up to eight additional voices from a sound source, with adjustable pitch, formant, panning, delay, filtering and modulation for each voice. This means you can employ Harmony for natural sounding harmonies or push out the boat for some creative voicings and epic vocal stacks. And the best thing about it is that all of this can be performed live in real-time, either in the studio or on stage.

Step by step 1. Instant accompaniment with Waves Harmony

1 Once Harmony has been added to a solo vocal track, the plugin will automatically process the audio, which is why you'll hear it pitch-correcting. You can choose the Key, Scale of your session, as well as choose from three pitch-correction modes: Natural, Hard and Notes.



2 Of course, we don't want to just use Harmony as a tuning tool – harmonies are what we're after, and there are three ways of getting them. You can auto generate notes, play them in using your MIDI keyboard, or click anywhere on the GUI to add a harmony.

3 By dragging notes around the GUI you position them in the stereo field and affect their level. With a note selected, you can transpose it across 48 semitones using the Pitch knob. Flat drops the pitch to a given note, while the treble clef button corrects it to your scale.

4 If your harmonies are a bit too bright, roll off some of the top end with the Filter, which you can set to either low-pass, high-pass or band-pass. The Delay slider will offset the harmony against the solo vocal in milliseconds, and can be further controlled with Feedback.

5 Clicking the Global button offers up an extra set of controls to affect all the harmonies. There's pitch and formant offset, Glide and stereo image spread. It's worth noting that you can also affect the balance of the source sound and harmonies if you want more of one.

6 If you're producing a complex vocal arrangement, you can capture snapshots of different harmony setups and then automate them as you move through the song. Modulation options are available in the form of an LFO and a Sequencer, which are assignable by drag and drop.

Step by step 2. Multiband delay with Lese Strum

1 Strum is a neat little multiband delay which grants you deep control over up to 15 bands of your audio signal. You can set the number of bands using the Bands dial, higher values will sound smoother but be more CPU intensive.

2 You can adjust the delay time according to frequency by dragging the red node in the spectrum window, and set the maximum delay time using the delay dial. In Free mode, the delay time is measured in milliseconds. In sync mode, the dial sets the maximum number of delay repeats.

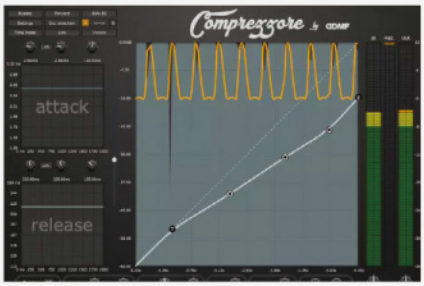
3 Click the blue feedback button to view the feedback control, where you can adjust your signal's feedback level according to its frequency using the blue node. You can also toggle low shelf and high shelf factor bands, and adjust the frequency, amplitude and bandwidth with the parameters to the right of that.

4 Add modulation to your signal with the Modulation dial, and adjust its speed with the Rate dial. You can also switch on skew, which adds or subtracts bias from the other modulation parameters dependent on the delay band's frequency. The modulation's flutter mode shape resembles a sine wave, and Random mode adds jitter.

Compression forefronts

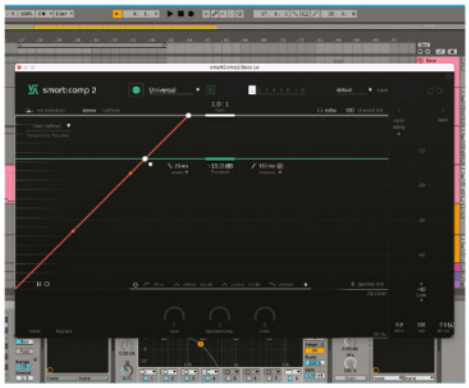
The compressor is a strange tool to consider. On one hand, its remit – grappling automatically with audio levels – is a pretty simple one. But once you consider how much potential this process has for imbuing sounds with personality, power and character, you start to realise there's a certain magic to compression. This magic is actually driving a lot of how we make music.

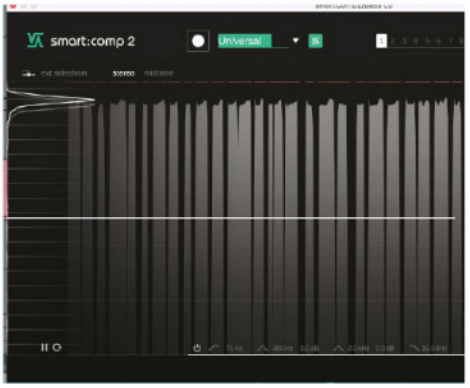
Innovations in compression in recent years have included more tailored sidechain functionality: Pulsar 1178, for example, with its full sidechain EQ and even delay. And then attack and release timing have started to crop up (as in smart:comp 2). Variable responses a la DDMF's Comprezzore have also been developed, and the subject of spectral compression looms large here as well. We certainly don't seem stuck for new fronts to explore in the compression landscape.

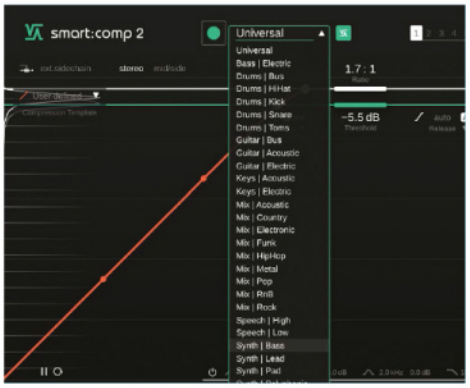


DDMF's Comprezzore goes further with both custom Knee shapes and attack and release flexibility

Step by step 3. AI compression with smart:comp 2







1 Sonible's smart:comp 2 has been making waves since its recent release, being praised for its intuitive design and ultra-clean 2000-band compression. However, as with most Sonible products, the most exciting feature is the AI that can give a good approximation of compression in seconds.

2 By clicking the record button at the top of the UI and playing your track, smart:comp 2 will "listen" to your track. After a few seconds the plugin will provide what it deems to be the best universal compression settings. smart:comp 2 aims for ultra clean compression, not aiming to add too much character.

3 The final step is to select a compression model. Open the drop down menu next to the record button and you'll find 31 different compressor models divided into categories such as Drums, Guitar, Keys, Bass etc. smart:comp 2 doesn't need to listen to your track again, so you can audition these models seamlessly.

GS DSP MagicBlur

As technology's power grows, so does its ability to process audio in new and exciting ways. Spectral processing is one of many relatively modern audio processing techniques that has been made possible by recent leaps and progress in computing power. The FFT-based technology can take many shapes, from EQs and audio restoration to reverbs and delays. Ultimately, it works by breaking down audio into tiny frequency bands which can then be individually targeted and treated. Although the process has been around for a while, it's only recently that consumer technology has been able to work with audio on a spectral level due to the CPU-heavy nature of the task.

Now, though, there are a growing number of software developers with practical audio applications for the technology. One such company is Gustav Scholda Digital Signal Processing. The Austria-based company specializes in spectral processing, with a catalog of five plugins that harness the power of this audio treatment approach. The range of products and their various purposes go some way to demonstrating the versatility of spectral processing, from clinical utilities to extreme creative sound design plugins.

GS DSP's capable spectral processing engine promises to offer innovative and practical approaches to carrying out common music production tasks including EQ, delay and pitch and frequency shifting.

> "Plate reverbs like the EMT 140 are not totally naturalistic in their sound"

Step by step 4. Spectral blurring with GS DSP's MagicBlur

1 MagicBlur is a unique spectral blur and freeze effect, capable of creating an array of tones and textures. The curve editor displays the blur time as a function of frequency, with the X axis representing frequency and the Y axis representing time. Select a curve by clicking on one of the horizontal colored lines.

2 You can drag the nodes to move the curve, or double-click to create and delete nodes. You can toggle between stepped curves and free curves by clicking on the top icon to the right of the input dial. For finer frequency adjustment, click the button below that to turn off the regular note type grid.

3 There are four selectable curves which you can alter independently, or select a curve preset by clicking on the grid icon. Once you have created your curves, you can morph between them using the X/Y controller to create seamless shifting in the effect.

4 MagicBlur's Invert, Shift and Skew dials let you quickly manipulate the frequency and time your curve is affecting. You can also set an offset between the left and right channels using the V dials below each of these controls, which creates a stereo spread effect.

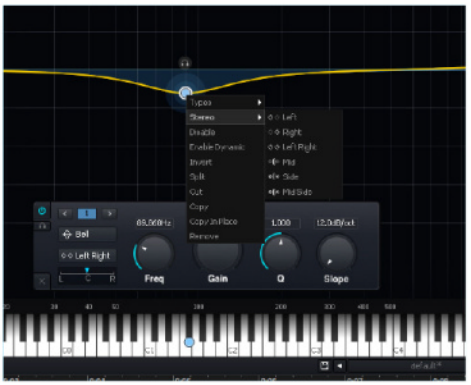
5 Add modulation by clicking on a grey circle below the Invert dial, then click on the triple arrow icon to select a destination. Modulatable settings are grayed out with yellow markers, simply click on a setting to assign the modulation to it.

6 There are twelve modulation sources to choose from, including custom LFOs, pitch followers and even a camera input. Click on the dropdown arrow to the right of LFO to select a modulation source. You can mix and match up to eight different modulation types for unpredictable evolving effects.

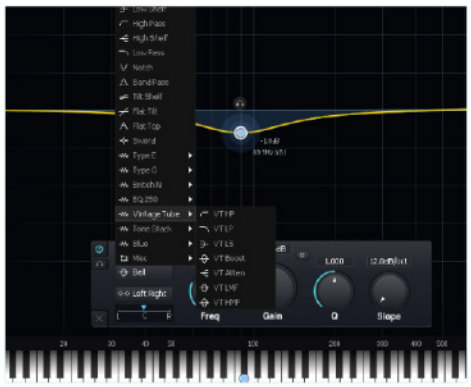
Step by step 5. Getting real with Kirchhoff EQ

01 Three Body Tech's Kirchhoff EQ is a 32-band parametric EQ, with nine vintage models, dynamic processing and a feature list that puts many more established EQ plugins to shame. At first glance, it resembles other EQs you may have used, but the power of this plugin is tucked away under the hood.

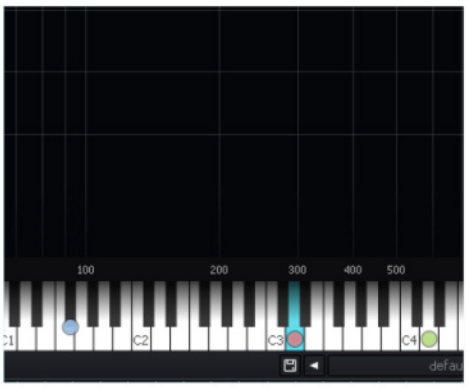
02 Double-click anywhere on the spectrum window to create a new band, which you can then drag to adjust its frequency and gain. Hold Ctrl/Cmd and drag to adjust a band's Q value, or hold Shift to make finer adjustments. Hold the headphone icon to solo an individual band.



03 Select a band to view its control panel. Here, you can change more advanced parameters, such as panning, stereo type, filter type and continuously variable slope. Alternatively, you can right-click on a band's node to access these settings, plus much more.

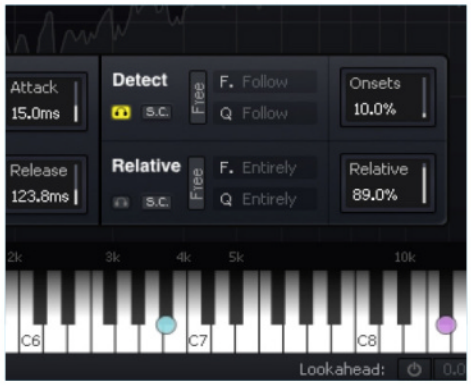


04 Kirchhoff has 15 distinct filter types to choose from, including nine vintage models that emulate some of the most desirable classic EQs including Neve, SSL and Pultec. There are high- and low-pass filters, shelf and bell shapes, plus 'sword' and brick wall cuts for precisely carving out very specific frequencies.



05 You can view the keyboard by clicking on the keyboard icon at the top left of the GUI. All bands will be displayed on the keyboard in the relevant color, and you can add new bands by double clicking on the respective note.

06 To access Kirchhoff's dynamic functionality, click on the D icon in a selected band's control panel. The Below and Above parameters dictate how the EQ responds to signals quieter and louder than the threshold respectively. You can turn off the Below or Above action by clicking on the relevant title box.



07 In Detect mode, the dynamics will be controlled according to the signal within that specific band. In Relative mode, the dynamics will be controlled by the entire audio signal, allowing for a more cohesive and balanced overall signal. You can solo each of these envelope detection modes to listen to what each of them is doing.

08 Using Kirchhoff's Free dynamics mode, you can manually configure a band's dynamic processor to respond to any frequency across the spectrum. Simply switch on Free in Detect or Relative mode, then drag the central node of the corresponding horizontal sliders. Drag the outside nodes to adjust bandwidth.

09 It's clear that Kirchhoff is an extremely comprehensive EQ with a huge arsenal of features. To make its usability as streamlined as possible, Three Body Tech has added a wide range of customization options within the settings menu. From here, you can set default filter behaviors, adjustment speeds, filter colors and even favorite filters.

Where next for synthesis?

On one hand, synthesis has forefronts in many areas of digital technology, but from another perspective, things can look a little unchanging. Even today's most interesting and innovative synths operate under a basic guiding principle: take a source sound (or multiple), and then process that in a variety of ways.

Really, the innovation in synths must come from one of these two areas: either in development of a better, more stunning or never-before-heard sound generation, or in more interesting, unique or exploratory ways of sculpting or enhancing that signal. Synths like Newfangled Audio's Generate or Physical Audio Preparation take their inspiration from real-world phenomena abstracted into a useable, programmable and, ultimately, loveable instrument.

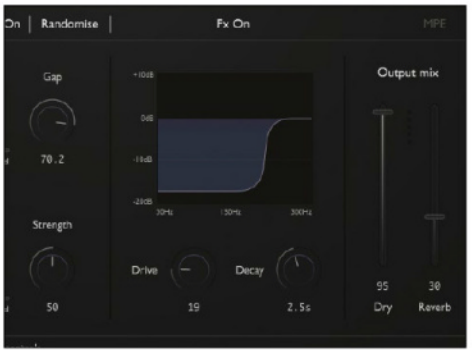
So where next for synthesis? We can't tell you exactly, but we're itching to find out.

Step by step 6. Physical modeling synthesis needs Preparation

01 Physical Audio's Preparation is a physical modeling synth that uses mathematical modeling to emulate the sound of two core sources: strings and rattle. Preparation shows a visualization of the modeling at the top of the GUI, which shows the two strings or bars and the rattle.

02 There are traditional attack and release controls to the right of the visualization, plus a tailboost control which increases the gain over several seconds. Below the visualization, you can select the String Setup or Rattle Control. The String Setup gives you independent controls of the top and bottom strings respectively.

03 In string mode, with the rattle off and inharmonics and nonlinearity turned down, the output is a relatively traditional string sound. From this blank canvas, you can start to tweak parameters to create more complex resonant tones. These settings are note-on, and take effect at the next note-on event.



04 Switch the Rattle On and enter the Rattle Control page to adjust the top and bottom rattle properties for even more advanced sound design. You can also assign velocity modulation using the slider to the left of each parameter. EQ, drive and reverb let you finalize your signal.

SOFTWARE TO WATCH

MINIMAL AUDIO
MorphEQ
$49
The mighty morphing EQ is all about modulation, taking its filter bands on rhythmic, programmable journeys and bringing the EQ to producers' consciousness as not just a mixing tool but a full-on creative effect – especially thanks to its series/parallel configuration options.
minimal.audio

MODALICS
Beat Scholar
$99
Circles divided into different numbers of parts help you get your head around rhythms in this plugin. Craft yourself some cyclical, divisible beats with the help of Beat Scholar's intuitive main window, and get yourself in the mood for pizza at the same time! Jokes aside, Beat Scholar is a compelling and interesting take on drums.
modalics.com

iZOTOPE
Neutron 4
$399
The AI-assisted mixing package has recently been boosted for version 4, and it was already pretty darn useful before that anyway! A high-quality plugin that can go as deep as you want, or simply provide you with quick results that are 95% of the way there
izotope.com

John Howes

The producer, developer and label head on the generative ideas behind new project Paperclip Minimiser.

Electronic musician, Cong Burn label head and developer John Howes' music lives on the boundary between chaos and control. On the one hand, his work is largely produced using systems and tools of his own design: rhythms and notes are generated using Strokes, a sequencing environment he's developed, synth patches are meticulously constructed on his Nord Modular G2 and Elektron Machinedrum, while parameters are modulated using another self-built software tool, the global modulation matrix Dispatch.

But though he's rigorous in his command of the creative process, Howes tells us, it's when he begins to let go of all this that the magic happens, through generative music-making and semi-autonomous agents that possess what he's decided to call 'lo-fi AI'.

His latest release, a self-titled project under a new alias, derives its name from Paperclip Maximiser, a thought experiment that envisions how an AI tasked solely with manufacturing paperclips could run out of control and inadvertently kill us all in the process, were it not instilled with some form of self-correcting, machine-based ethics.

Thankfully, the AI-powered tools used to produce Howes' latest release, Paperclip Minimiser, haven't yet threatened us with extinction: instead, they've helped him record some of the best experimental music we heard throughout the whole year.

How did you get into music-making?

I started making tunes on PlayStation 1 and cracked software, FL Studio on PC. Then, when I moved to Manchester at the age of 18 I got into a bit more music and started buying equipment – drum machines and effects and modular stuff, and now I'm kind of in between all of it. I'm still using all these things – not so much cracked software, though [*laughs*]. But still torn between the world of drum machines, synths, hardware, and software.

When did you move from playing around to actually working in development?

When I was at uni I made a bunch of Max for Live devices. I had one of those Eurorack converter things where you can send CV out. I didn't have a lot of money, as a student, so I was making LFOs and sequencers in Max that would output CV into my modular, basically as a way of saving money. And then I didn't touch it for five or six years.

I worked in the music industry for five years, and I got to the stage where I was like, the next job that I get in this industry is going to be working on stuff that I don't give a shit about. Mainstream, money-making, business techno or pop music type of thing. So it was like, I guess I'm at the end of my road here.

I just left the music industry completely, didn't put out a record or do any work on music for about a year, and studied coding. Then, the first job that I got in coding was for Behringer. So I kind of went straight back in.

Tell us about Strokes, the new plugin you've developed?

I was reading a book about Jaki Liebezeit, the drummer from Can. He's an absolutely sick drummer, and he's got this really repetitive style, with really complex rhythms. I wanted to make music that sounded like that, but I didn't want to program it in MIDI clips in Live, it'd take fucking ages. So I started looking at these ways of devising rhythms and came across Euclidean rhythms. When you look at the stuff Jaki was doing, he never says the word Euclidean, but it's all there – all of it is about mathematical relationships, and the diagrams he's drawn are Euclidean rhythms, he just didn't call them that. So I started working on it mainly as a way of generating interesting drum patterns that constantly evolve.

Before I started working on it, I used to be into Eurorack stuff, I had this big Eurorack system made of all Doepfer stuff. I used to build mini versions of Strokes on that. I made a hardware version, years and years ago, and had this whole system patched for a year, and was constantly generating stuff with it. Then I turned it into a Max thing, later on, and it's been in development for nearly four years now. It started as a small thing to generate rhythms, and it's grown into a full-on sequencing environment. Pretty much all the music that I've made in the past three or four years has started with it in some capacity.

Tell us about the new record. What's behind the decision to release under a new alias?

It's quite different to a lot of the previous things that I've done, in terms of process. The reason that I put it under this new name is that, at the core of all the tracks on this new record, I'm into the idea of trying to put semi-artificial intelligence into the generative system. Most of the music that I make comes from complex generative systems made using Strokes, Dispatch and other bits of equipment. But in the techniques that I use on this record, the focus was on trying to give each voice or each section of a track its own space, almost like it's an actor in a play, or something like that. Almost like Curb Your Enthusiasm ambient music, where everyone knows where it's going, but as soon as someone actually hits Play on it, you don't know exactly what the results are gonna be.

The idea is that all the instruments, all the voices and all the parts all listen to each other and respond to what each other are doing. But also they have to have their own intelligence in a way, they make their own decisions. That's where the name comes from. It's called Paperclip Minimiser. Paperclip Maximiser is this thought experiment about an AI that is given the job of making paperclips but ends up destroying the universe. So this is kind of a self-deprecating name, where it's struggling to make ambient music. [*laughs*]

You've mentioned that you mostly used the Nord Modular G2. What is it about this synth that you found inspiring?

Oh, man. The Nord lets you design your own drum machines, synths, effects and sequencers. It's almost like a VCV rack, but the modules haven't been updated for 15 years. With the Nord, I get the sense that they threw everything they had into it. All the best engineers were working on it, and they basically went bust trying to make it. The amount of engineering expertise that went into this machine make it one of the best machines out there. All of the oscillators, all of the tools, and everything sounds great.

There's just enough in there that you could spend your entire life digging into these things. Now that I've got it, I kind of don't want to buy any more equipment ever again – every synth that I want, I can just try and make it on the Nord. So I've got a Lyra-8 that I made in there, and I'm gonna make a Bastl softPop tomorrow. A lot of the time a new synth'll come out with a technique in it, but rather than having to spend any money and buy it, I can try to build a close approximation on the Nord and usually end up somewhere else that I didn't expect.

I bought it as a prototyping environment, and that's the reason why I spent two months learning it. And I still know 10% of what that machine is capable of. It's insane. I wouldn't recommend that everybody pick one up because they're really expensive and also really temperamental, but yeah – holy shit, man, this machine. It's the end game now.

You've also mentioned both the Machinedrum and Monomachine...

The Monomachine and the Machinedrum, I didn't use a ton. I didn't use any of the internal sequencers on either of them. A lot of what I was doing was building sequences on the Nord and sending that out to the Machinedrum and Monomachine. The Machinedrum, I've had a couple of them over the years, and they're amazing. It's one of those things where I get one and I'll use it for a year, get sick of the sound of it, and sell it or put it on the shelf.

I'm not as interested in the new stuff, but the Monomachine, and the Machinedrum especially – it's basically a modular synth. You can do a ton of internal routing stuff.

There are infinite things that you can potentially do. It's similar to the Nord in that it's an open box and you can do what you want with it. Simple tools that you can patch together in really crazy ways. I'm not as deep into the Monomachine, I'm not as keen on the synth engines. I don't use either that much anymore, to be honest, since I've got the Nord. I've got close approximations of all the Machinedrum and Monomachine voices that I've built myself inside the Nord. Other things are slowly getting pushed out. Maybe in 10 years or so, it'll just be 10 Nord Modulars. [*laughs*] ■

JOHN HOWE'S TOP TIPS

Don't listen to any music

1 Disengage with music, or try to listen to wildly different music while you're working on stuff. No Boomkat, no social media, no Discogs, no listening to what's going on and no going to clubs.

Take a more disciplined approach to your work

2 I used to make music as a hobby, where I'd finish work for the day, turn on the machines and jam for a couple of hours. Most days I'd be tired and make something not very good and get a bit upset with myself and stop making music for a bit. Now I'll not make music for three or four months, but build up a big list of technical or conceptual ideas to try out later. I'll write them down, let them mutate into new ideas, and next time I go to make music I've got some new tasks I want to achieve.

Be realistic

3 This sounds like some self-help thing, but it's not at all. If I've got three or four hours and I'm fresh, I'll work on mixing or sound design, maybe build a new patch or work on arrangement. These are gritty tasks that aren't very fun but require a lot of concentration. But if I've been at work all day, it's fine to just put some white noise into a filter and move the filter around for an hour. That's fine too. We've got to take the realistic amount of time and energy available and still make something useful out of it.

GrainLoop

Isotonik Studios' new granular loop app destroys your loop to save it.

By Michael Ross

Looping your guitar is a great way to create ambient pads or rhythmic backdrops. But loops, by their nature can be static – the same thing going round and round. One way to shake things up is granular looping. A granular looper slices your loop audio into tiny bits, and then rearranges and sometimes reverses them. In the world of iOS there are apps like iDensity and Borderlands. In the plugin world there is Stream, Spacecraft and others. To this latter list, now add GrainLoop.

GrainLoop triggers a continuous stream of sound particles and rearranges them in time to mangle the original loop. Developed by Javier Salthú, aka NOISS COKO, and published by Isotonik Studios, it offers real time audio recording and unlimited overdubbing, as well as Random and LFO modulation for Point, Speed, Density, Detune, Pan and Level.

In GrainLoop the recording/overdub process doesn't occur on the same buffer used for playback. The plugin uses separate buffers for each action. While this prevents clicks and unwanted noise, it doesn't allow real time processing. When you engage the Rec mode the recording starts recording but the recorded sound is not heard. Once you then turn Record off, the buffers crossfade and the new layer is in playback.

By leaving Play and Record on, I could have GrainLoop continually, silently record an adjustable length of audio that would play as soon as I turned record off. This is reminiscent of the prized, original EHX 16-second looper, that was also always recording. Hit record again and more sound layers are overdubbed. The buffer can be emptied by clicking the Clear button.

> "Playback Direction Switches between forward and reverse. Length defines a buffer size or loop period between 1 and 30 seconds."

Switching on the Freeze toggle causes the small loop fragments to be played back but over a fixed location in the timeline. The Point dial can be used to scan through the sample manually. This could be controlled by a MIDI expression pedal or knob for some cool effects. Playback Direction Switches between forward and reverse. Length defines a buffer size or loop period between 1 and 30 seconds. Unfortunately, unlike with Ableton's Looper, if you set a new length while a loop is recorded it will immediately clear the buffer of all its previous recordings. GrainLoop also offers pitch control. I could reduce the density and speed of the granulation and have backwards octaves randomly pop in to my solo (albeit still slightly mangled), or record harmonized EBow lines.

Further manipulation is made possible by Randomize and LFO modulation sources with individual control over the modulation amount applied to each parameter. The Save button allows you to choose a specific folder for every new recording to be automatically exported (to, for example, the Samples/Recorded folder in the session) and recalled when the Live session is reopened. Once a path is assigned, a different sound file will be exported every time a new layer is added to GrainLoop's buffer.

There's more on the Isotonik Studios site, but you can probably already tell that Salthú has come up with a worthy addition to the looper family. GrainLoop has its own thing going and is worth checking out for either real time performance or recording. ■

WAVEFORM FREE 12

"All in all, Waveform Free is so capable that you'll wonder why you didn't have to pay for it..

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